Soundcraft TS12 USER MANUAL

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Note.

CPS 450 or 650 Power Supplies have a seperate USER MANUAL.

1.00 SPECIFICATIONS

Distortion	- Any signal path me	asured at +1	0dBu	1kHz:<0.005% 10kHz:<0.01%
Crosstalk	- Input to any unrou Between groups at		kHz	-90dB -80dB
Noise	 DIN audio bandwidt Mic input EIN, 200 Typical output noi 	Ohm source	els route	-128.5dBu d -82dBu
Frequency response - Measured at +10dBu, equaliser bypassed Ref 1kHz				
	KCI TKIIZ	20Hz	:-0.5dB,	20kHz:-1.0dB
Gain	- Maximum mic gain Line to group outpu	t		90dB 30dB
Operating	levels - External int Internal lev			+4dBu -2dBu
Input and		Mic Line level i Any output	nputs	2kOhm 10kOhm <75Ohm
Maximum output levels - Unbalanced outputs +21dBu into 5kOhm Balanced outputs +26dBu into 6000hm				
Dimensions	- 24CH frame (std pa 24CH frame (metal 36CH frame	-	L:2170,D:	910,H:1090mm 910,H:1090mm 910,H:1090mm
	Height at faders			760mm

2.00 GENERAL DESCRIPTION

The Soundcraft TS12 is a compact in-line console based around TS24 technology. The TS12 offers quality and sophistication as well as presenting the user with a clear and easy to use layout.

The TS12 utilises the Mix and Channel concept, as found in the TS24, whereby, the path designal CHANNEL always goes to the routing matrix, whilst the MIX signal path normally goes to the Mix bus. Thus, your monitor mix in record becomes a rough mix for mix-down.

Normally the TS12 has 6 auxiliary sends, however, with a combination of a "BOUNCE" facility and "FDR", an optional 4 stereo auxiliary sends are made available during mix down.

All inputs and outputs are electronically balanced. Electronic balancing reduces the signal degradation introduced by more conventional transformer coupled designs, and ensures superior transient response, minimal phase shift, and excellent common mode rejection even at high frequencies.

Semi-parametric Equalisation, which may be used in either the Channel or the Mix path, provides exceptional musical enhancement and, when used in conjunction with the high pass filter, almost total control over much of the audio spectrum.

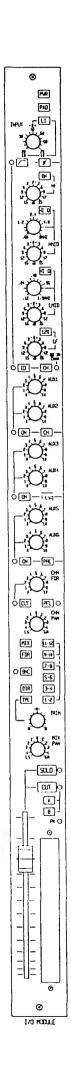
A patch bay provides standardised connection points to manu sections of the console circuitry and to numerous pieces of external equipment. This allows rapid insertion of peripheral equipment into the system by means of patch-cords.

The TS12 is entirely modular, giving great flexibility in the selection of facilities. Standard configurations are 24, 36 and 40 channels, with two patchbay options. The compact console configuration offers 24 I/O modules, the master section and 6 stereo FX/groupmodules, with a 9" bantam jack patchbay. Larger configurations can hold 24, 36 or 40 I/Os, with a more comprehensive 19" patchbay. The 24 and 36 sizes can have up to 4 quad return modules fitted, giving a full 16 effects returns.

DISC-BASED AUTOMATION

The TS12 automation package is a sophisticated disc-based system that combines SMPTE capability with a quick and easy means of control.

The real-time computer system controls channel fader settings, muting, EQ in/out and auxiliary on/offs. Each function on each channel can be individually set to read/write via the computer keyboard. A high-resolution colour display with instant pop-up menus allows an automation control function to be instantly selected. Any number of VCA sub-groups can be set up on the computer and all associated fader movements stored and recalled as with normal fader operation.



2.01 INPUT/OUTPUT MODULE

1. Input to channel

The Channel can be operated in either Microphone or Line mode.

The Channel Input is an electronically balanced design, configured for optimum low noise performance. The Microphone Input impedance is approximately 2kOhms, which provides optimum load for most studio microphones.

The high level Line Input impedance is greater than 10kOhms.

a) + 48

Capacitor microphones can be powered by the internal +48V phantom power supply by pressing the +48 button.

b) LINE

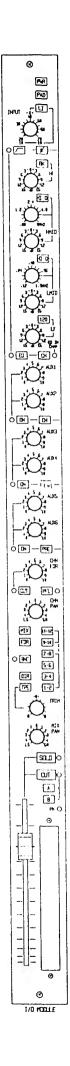
The LI button switches the input to receive a line level signal normalled from the patch bay and varelco tape interface.

c) PAD

Pressing the PAD button inserts a 20dB attenuator into the input of the microphone amplifier allowing high level input signals to be catered for, without exceeding the headroom capability of the console. Such signal levels can easily occur from high output capacitor microphones used in close proximity to a sound source.

d) PHASE

The phase of the input signal can be reversed by pressing the PHASE button to correct for mis-wired microphones or out of phase microphones in multimic set ups. It can also be used to correct phase reversal on line inputs to the console.



e) I/P GAIN

the Input Gain can be varied between +22dB and +70dB of Gain, using the Input Gain control. Used in conjunction with the 20dB Pad a 50dB control range is available. When line is selected the input signal can be adjusted within a range of -10dB to +14dB.

2. Equaliser section

A 4 band semi-parametric equaliser is provided with bandwidth control on the middle frequencies.

a) High Pass Filter

The High Pass Filter has a fixed turnover frequency of 80Hz, below which the signal is attenuated at a rate of 24dB/Octave and can be switched in or out independent of the equaliser using the FIL button.

The position of the High Pass Filter in the signal path is determined by the CH button. (See h).

b) HF (High frequency)

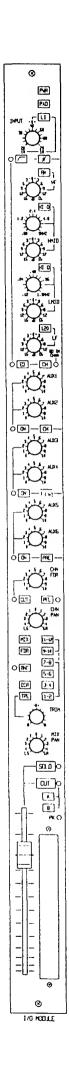
The frequency is switchable between 8kHz and 12kHz. 15dB of boost or cut is available, with a "shelving" characteristic. Ie. The slope of the curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

c) HI MID

The frequency is continuously variable between 600Hz and 9kHz with 15dB of boost or cut, with a peak/dip characteristic. The bandwidth (Q) of the filter can also be switched between Q=0.5 or Q=1.4.

d) LO MID

This is identical to the Hi Mid frequency except that the frequency is continuously variable between 120Hz and 1.9kHz. The bandwidth (Q) of the filter can also be switched between Q=0.5 or Q=1.4.



e) LF (Low frequency)

The frequency is switchable between 50Hz and 100Hz. 15dB of boost or cut is available with a shelving characteristic.

f) EO

The Equaliser can be switched in and out of the signal path, independent of the High Pass Filter, by pressing the EQ button. The associated LED will light when the Equaliser is in circuit.

g) CH

When CH is press the High Pass Filter, Equaliser and insert point are placed in the channel path, however, the High Pass Filter is not activated until FIL is selected and the Equaliser is not activated until EO is selected.

4. Auxiliary section

There are 6 Auxiliary sends available in pairs for use as echo, foldback or other auxiliary effects units.

AUX 1-2

a) LEVEL CONTROL

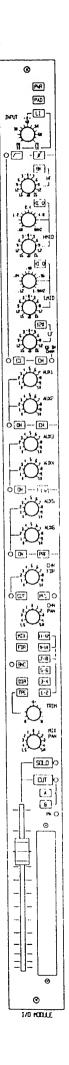
The Aux 1 level control is located above the Aux 2 level control.

b) CH

Normally all six Aux sends are fed from the Mix path. By pressing the CH button Aux 1 & 2 sends may be taken from the Channel path to allow the headphone mix to be derived from individual inputs or the recorded signal, which may include external effects. If FDR and CH are pressed then Aux 1 & 2 sends become post mix fader sends.

c) ON

Enables the pair of auxiliaries.



AUX 3-4

a) FLW

When the FOLLOW button is not pressed Aux 3 and 4 are fed by the post Mix fader signal.

When pressed Aux 3 and 4 follow the signal in Aux 1 and 2. This can be either pre Aux 1-2 level or post depending on jumper selection.

JUMPER SELECTION

Place jumper J2 ON to take the signal for Aux 3 from before Aux 1 level.
Place Jumper J3 ON to take the signal for Aux 3 from after Aux 1 level.
Place jumper J4 ON to take the signal for Aux 4 from before Aux 2 level.
Place jumper J5 on to take the signal for Aux 4 from after Aux 2 level.

b) ON

Turns Aux 3 and 4 on.

AUX 5-6

The Aux 5 level control is located above the Aux 6 level control. Aux 5 & 6 are always driven from the mix path.

a) PRE

Selecting PRE takes the signal from before the mix fader.

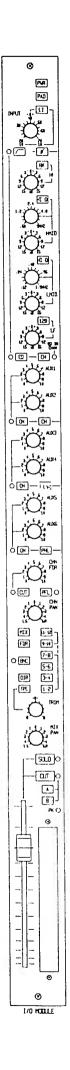
b) ON

Turns Aux 5 and 6 on.

5. Channel Fader section

a) CHANNEL FADER

The Channel fader is a precision rotary control used to set the level to tape during recording. It can also be used as an Aux send level control or return level control during mix down.



b) CUT

The CUT button mutes the Channel Path with the exception of Aux 1 & 2.

c) PFL

Pre-Fade Listen soloes the pre-fader, post insert signal, independently of the CUT button. PFL operation is indicated by an LED on the Channel, and a master warning LED on the Master Module.

d) CHANNEL PAN

The Pan Pot is a centre detented control, with a loss of 4.5dB at the centre point. This is a satisfactory compromise between the 3dB loss required for constant power panning, and the 6dB loss required for constant voltage panning.

e) ROUTING

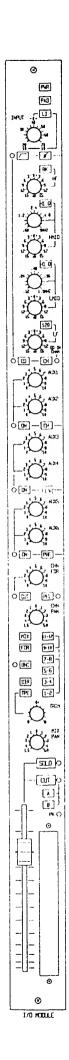
The signal can be routed to any or all 12 Groups or the Stereo Mix bus by pressing the relevant button and panning hard left for odd numbered groups, or hard right for even numbered groups.

f) FDR

The FDR button feeds the signal present in the MIX path (post fade/pre pan) to the Channel Fader and Pan Pot, OVERRIDING the normal input. This allows the channel fader and Pan Pot to be used as an extra send from the Mix path, thus stereo Aux sends via the routing matrix are possible during mix down.

q) BNC

The BOUNCE (BNC) button allows simple track bouncing to be performed. When BNC is selected the Mix fader and its associated Pan Pot are disconnected from the main Left and Right buses and sent to the routing matrix, OVERRIDING the Channel Input. Thus a sub-mix can be balanced on the long throw Mix faders and can be transferred at once to any selected pair of tracks without having to try and duplicate this balance on the rotary faders and pan pots.



When FDR is used in conjunction with BNC, routing to groups 9-12 and the mix bus are fed by the output from the channel pan pot and routing to groups 1-8 are fed by the output from the mix pan pot. This gives a total of 6 mono and 4 stereo auxiliary sends and 4 stereo sub-groups during mix down.

h) DIR

This button routes the output of the channel directly to the respective Tape Send AND completely OVERRIDES anything else assigned to that particular output. If FDR is selected the direct signal is the pre-fade channel signal.

i) TAPE

Normally the group or direct signal drives the mix path and metering. Pressing TAPE replaces it with the tape return, overriden by patching into the MIX IN jack.

j) TRIM

Gain trim for the mix path input giving +/-8dB of adjustment. This does not affect the signal to the meters.

6. Mix Fader section

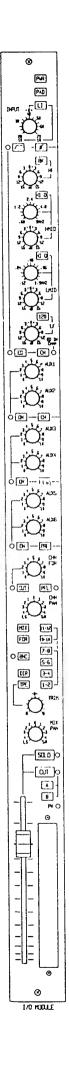
The Input to the Mix path can be either Tape Return or Tape send (which can be either the main groups or the direct output from the channel).

a) MIX PAN

The Mix Pan Pot is centre detented with a loss of 4.5dB at the centre point.

b) SOLO

Solo follows the set up on the Master module, (either solo-in-place or PFL). If solo-in-place is selected then the soloed signal will appear in its normal stereo position. If PFL is selected then a mono feed will be heard in the control room rather than the main mix.



Jumper J1 needs to be installed for a module to be cut when SIP is selected. This is normally fitted as standard and would be removed for any I/O used as an effects return, thus providing a SOLO SAFE facility.

c) CUT

Provides a silent FET mute of the mix path.

d) A and B

The A and B buttons allow the module to be programmed to mute the mix path when the Master Mute buttons are pressed on the Master module. Ie. If A is pressed on a module, that mix path will be cut when master cut button A is pressed on the Master module.

e) PEAK

The Peak LED inspects BOTH signal paths (Channel and Mix) in the module at the following points:Post channel input amplifier, post insert return and post equaliser. When any of these points exceed +15dBu (8dB below clipping) the LED will light. Even short duration transients will register since the peak detector has a fast response time but slow decay time.

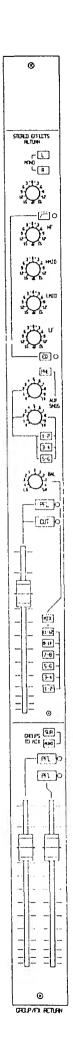
f) MIX FADER

The output of the Mix path is controlled by a high quality long throw fader. The law has been specially tailored to allow fine resolution in the upper working region, infinity cut off is greater than -90dB.

1) METERING

Metering for the tape sends/returns is provided by VU meters in the overbridge.

The Meter always monitors the module mix path signal. When the module is not switched to tape the meter will read the group output (tape send). When the module is switched to tape the meter will monitor the tape return level.



2.02 TS12 STEREO FX RETURN AND GROUP OUTPUT

The Group/FX return module provides final control over the group output level and allows the use of audio sub-grouping. The stereo FX return can be used in either mono or stereo mode and provides a stereo input to the console which may be routed to any or all of the group and mix buses.

FX RETURN

1. Input section

a) L and R

These switches allow the module to work in one of three modes.

- i. With both switches out the module works in stereo mode.
- ii. With either L or R in, both paths of the module are fed by either the left input or the right input.
- iii. With both switches pressed, both paths of the module are fed by a mono sum of the left and right input.

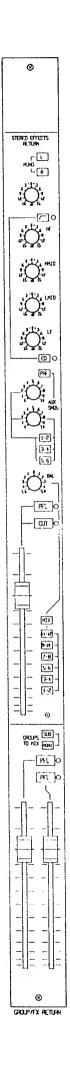
b) Input Gain

The active Input Gain control can provide +/-15dB of trim to the incoming nominal line level input signal.

2. Stereo Equaliser

a) High Pass Filter

The High Pass Filter operates at 100Hz with an ultimate slope of 12dB/Octave. This can be switched independent of the Equaliser by pressing the FIL button.



b) HF (High Frequency)

The frequency is fixed at 10kHz with 15dB of boost or cut available with a "shelving" characteristic. i.e. The slope of the eq curve does not keep rising with frequency, but having reached the desired amount, flattens out or "shelves" from that frequency on.

c) HI MID

The frequency is fixed at 3.2kHz. 15dB of boost or cut is available with a peak/dip characteristic. i.e. Having reached a maximum (or minimum in the case of cut) at the selected frequency, the response returns to zero on either side of that frequency. The shape of the curve when plotted gives the characteristic "bell" shape.

d) LO MID

The LO MID section is identical to the HI MID section, with the exception that it operates at 320Hz.

e) LF (Low frequency)

The frequency is fixed at 100Hz. 15dB of boost or cut is available with a "shelving" characteristic.

f) EQ

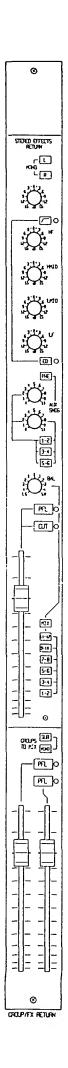
The Equaliser can be switched in and out of the signal path, independently of the High Pass Filter, by pressing the EQ button. The associated LED will light when the Equaliser is in circuit.

3. Auxiliary Section

The auxiliary sends can be routed to any or all of the 6 sends in pairs and can be selected pre or post the fader.

a) PRE

Takes the feed for the send control from before the FX return fader.



b) Level controls

The top potentiometer controls odd numbered sends and the bottom controls the even numbered sends.

c) Routing

The signal can be routed to any pair of six auxiliaries.

4. Channel output section.

a) Channel Balance

The Balance control is a centre detented potentiometer and corrects or deliberately creates any errors in stereo imaging.

b) PFL

Pre-fade Listen soloes the pre-fader, post insert signal, independent of the ON switch. PFL operation sends a mono feed to the control room monitor (L and R sum). It is indicated by a red LED on the channel, and a master warning LED on the Master Module.

c) CUT

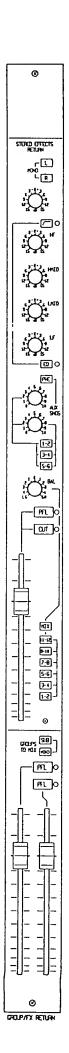
The CUT button mutes the Channel Path

d) ROUTING

The stereo signal may be routed to any or all 12 Groups and Stereo Mix by pressing the relevant button.

e) CHANNEL FADER

The Channel fader is a long throw fader giving good resolution and infinity cut off greater than $-90\,\mathrm{dB}$.



GROUP OUTPUT

The group output section is provided with a pair of faders, individual PFL sends and sub-grouping facilities. Insert points are also available after the group summing amp.

a) SUB

Pressing the SUB button allows you to route a pair of groups directly to the stereo mix bus. For audio sub-grouping while in mix down the group faders can become the sub-group masters.

b) MONO

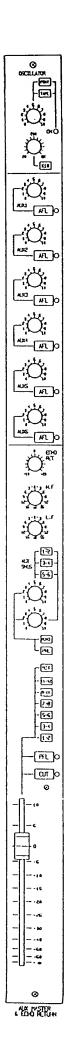
Sends both group output signals to the left and right buses.

c) PFL

Pre-fade Listen soloes the pre-fader, post insert signal, independent of the ON switch. PFL operation sends a mono feed to the control room monitor (L and R sum). This is indicated by an LED on the channel, and a master warning LED on the Master Module.

d) Group Fader

The group faders are long throw faders giving good resolution and infinity cut off greater than -90dB.



2.03 AUXILIARY/ECHO RETURN

The Auxiliary/Echo return module provides master level controls for the six auxiliary sends and the controls & equaliser for the stereo echo return. There is also an oscillator for tape alignment and test purposes.

1. Oscillator

a) PBAY

Pressing PBAY brings the oscillator up as an output on the patchbay

b) TAPE

Pressing TAPE routes the output of the oscillator to the 12 groups and the stereo mix bus.

c) LEVEL

Adjusts the level of the oscillator.

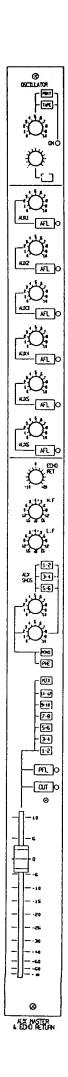
d) FREQUENCY

Frequencies available on the oscillator range from 20Hz to 2kHz. When the X10 button is selected the range is 200Hz to 20kHz.

2. Auxiliary Masters

There are 6 auxiliary send master level controls. Each master has its own AFL, after-fade listen, facility. The LED illuminates when AFL is pressed.

When AFL is selected the current signal in the control room monitors is replaced by the AFL signal.



3. Echo return

a) Level control

The active input gain control can be adjusted between -10dB and +20dB in order to take advantage of the headroom and noise performance.

b) Stereo Equaliser

A dual band stereo equaliser is provided with shelving characteristic. The High frequncy turnover is at 8kHz and the Low frequency turn over is at 70Hz.

c) Auxiliary sends

The auxiliary sends can be sent to any pair of 6 auxiliaries and can be selected pre or post fade.

i Routing

The signal can be routed to any or all of the six auxiliary pairs.

ii Level controls

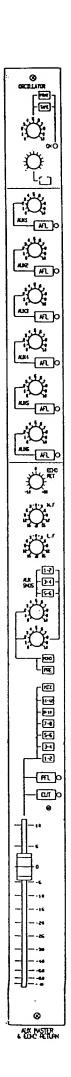
The top potentiometer controls odd numbered sends and the bottom controls the even numbered sends.

iii MONO

Pressing mono sends both left and right signals to both auxiliaries. Without mono selected the left channel feeds odd numbered auxiliaries and the right channel feeds even numbered auxiliaries.

iv PRE

Takes the feed for the pair of auxiliaries from before the echo return fader.



d) ROUTING

The stereo signal may be routed to any or all 12 Groups and the Stereo Mix by pressing the relevant button.

e) PFL

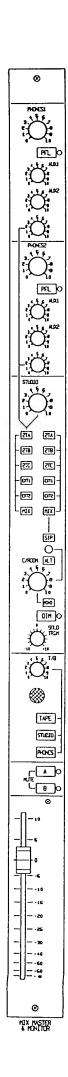
Pre-fade Listen soloes the pre-fader, post insert signal, independent of the ON switch. PFL operation sends a mono feed to the control room monitor (L and R sum). This is indicated by an LED on the channel, and a master warning LED on the Master Module.

f) CUT

The CUT button mutes the echo return

e) ECHO RETURN FADER

The Echo return fader is a long throw stereo fader giving high resolution and infinity cut off greater than -90dB.



2.04 TS12 MONITOR MASTER MODULE

1. Headphones Section

There are 2 stereo headphone outputs provided on the TS12 console, each with a master level control and PFL facility. The source to each may be any mix of Aux 1, Aux 2, plus a stereo mix selected by the studio monitor selector.

a) AUX 1&2

The Aux 1 and 2 controls add post Aux master signals to the headphone mix. The feeds are post patchbay insert so any other signal can be patched into the headphone circuits.

b) STUDIO

This control can be used to introduce the Studio Monitor signal into the Headphone mix.

2. Monitor section

a) STUDIO LEVEL

The signal present on the Studio Monitor selector can be sent to a Studio loudspeaker via the STUDIO level control.

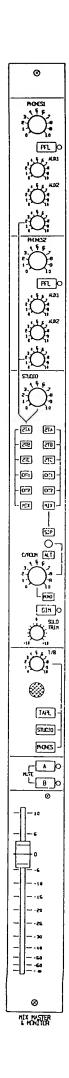
b) MONITOR SOURCE

Two sets of monitor source switches are available, one for the studio and phones 1 and 2 and the second for the Control Room Monitors or an alternative set of monitors.

The source for the Control Room Monitor or Studio can be either Main Mix left and right, or one of 5 external sources: replay from tape machines a,b,c, Ext 1 or Ext 2.

c) SIP

Pressing SIP (solo-in-place) brings any soloed signal up on the Control Room Monitors in its stereo position, all other Mix outputs from non-disabled I/Os will be cut.



d) ALT

An alternative set of loudspeakers may be fed from the TS12, these are controlled by the CRM level control and are switched in by pressing ALT.

e) CONTROL ROOM LEVEL

This is the master level control for the control room monitor.

f) MONO

This sums the left and right signals and provides a mono compatibility check on the control room monitors.

g) DIM

This attenuates the signal to the monitor by 20dB.

h) SOLO TRIM

Solo Trim allows you to adjust the level of any soloed signal to match the normal monitor listening levels.

3. Communications signal

a) TALKBACK

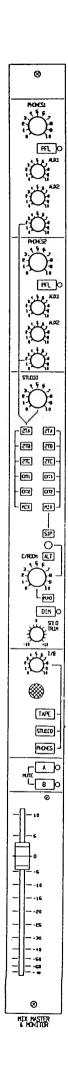
The Talkback Mic is an electret microphone and has a variable gain controlled by the Talkback Level Pot.

b) TALK TO TAPE

Routes the output from the Talkback microphone to all groups and the mix bus, thus enabling floor counts to be put on tape. A 30Hz slate oscillator is summed with the talkback for easy location on fast wind.

c) TALK TO STUDIO

This allows the Talkback microphone to be fed to the Studio Loudspeaker. The control room monitor is dimmed to prevent feedback, the switch has a momentary action.



d) TALK TO PHONES

This allows the Talkback microphone to be fed to the 2 Headphone circuits. The Control room monitor is dimmed and the switch has a momentary action.

4. Electronic muting

On pressing master mute A or B any pre-programed mutes set up on the I/O modules will be activated.

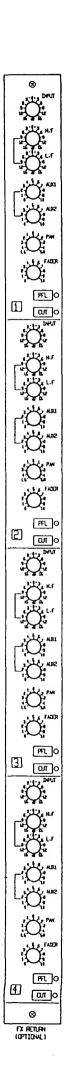
5. Stereo Fader

The Main stereo fader is a high quality long throw fader which has +10dB at the top of its travel.

6. Metering

The main Mix outputs are provided with wide scale VU meters.

The input to the Phase correlation meter is derived from the left and right master meters which follow the control room monitor selection.



2.05 OPTIONAL EFFECTS RETURN MODULE

The optional Effects Return module contains 4 individual Effects Returns with Auxiliary sends and two band EQ. All four FX returns are identical in function and circuitry. The Effects Return signals are always fed to the main mix bus.

1. Input to channel

The Effects Return module has an electronically balanced input with rotary active gain control to optimise headroom and noise.

2. Equaliser section

The equaliser is positioned in the channel path before the Auxiliary sends.

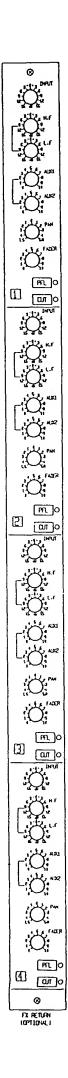
- a) HF (High Frequency)
 15dB of boost or cut is available at
 10kHz. The filter has a shelving
 characteristic, ie. the slope of the EQ
 curve does not keep rising with frequency,
 but, having reached the desired amount,
 flattens out, or "shelves" from that
 frequency on.
- b) LF (Low Frequency)
 15dB of boost or cut is available at
 100Hz, with a shelving characteristic.

2. Aux Send

The signal in the channel can be routed to either Aux 1 or Aux 2. Both auxes are fed from an internal jumper to select pre or post fade.

3. Pan Pot and Channel status

a) PAN
The Pan Pot is centre detented, with a loss of 4.5dB at its centre point.



- b) PFL Pressing PFL soloes the pre-fade signal, independent of the CUT switch. When pressed the associated LED illuminates, the signal is routed to the Control Room monitors and to the master meters.
- CUT An individual FX return is CUT when the CUT button is pressed. This mutes both the send to the Mix bus and the Auxiliary sends.
- d) FADER The individual fader is a rotary control.

NB: The four FX returns are always routed to the main mix bus.

3.00 Installation

3.01. Connector Conventions

All XLR type connections are normally wired to the following standard:-

Pin 1 GROUND

Pin 2 HOT (In phase signal)

Pin 3 COLD (Out of phase signal)

All inputs and outputs are electronically balanced.

3.02. General Wiring Procedures

To take full advantage of the excellent signal to noise ratio and low distortion of Soundcraft consoles care must be taken to ensure that incorrect installation and wiring does not degrade the performance of the desk. Hum, buzz, instability and Radio Frequency Interference can usually be traced to earth loops and inferior earthing systems. In some areas, especially heavily industrial areas, the incoming mains earth will not be adequate, and a separate technical earth for all the audio equipment must be supplied. However, check with your local electricity supply company to ensure that safety regulations are not infringed or negated.

The successful, hum free, installation of a system requires forethought, and the establishment of a set of ground rules, which must be consistently adhered to at all stages of installation.

1) Initial Wiring Considerations.

a) For optimum performance, it is essential for the earthing system to be clean and noise free, as all signals are referenced to this earth. A central point should be decided on for the main earth point system, and all earths should be "star fed" from this point. It is common electrical practice to "daisy chain" the earths to all electrical outlets but this method is unsuitable for audio installations. The preferred method is to run an individual earth wire from each outlet, back to the "system star" point to provide a safety earth of screen reference for each piece of equipment.

A separate earth wire should also be run from each equipment rack and area, to the "star point". This may or may not be used depending on circumstances, but it is easier to install in the first place, than later when problems arise.

- b) Install separate "clean" and "dirty" mains outlets, wired individually back to the incoming mains distribution box. Use the "clean" supply for all audio equipment and the "dirty" supply for all lighting, vending machines etc. Never mix the two systems.
- c) If necessary, to provide sufficient isolation from mains born interference, install an isolating transformer for the "clean" supply. The isolation transformer should be provided with a Faraday Shield which must be connected to earth.
- d) Never locate the incoming mains distribution box near audio equipment, especially tape recorders, which are very sensitive to electro-magnetic fields.
- e) Ensure that all equipment racks are connected to earth, via a separate wire back to the "star point".
- f) Equipment which has unbalanced inputs and outputs may need to be isolated from the rack to prevent earth loops.

2) Audio Wiring

Having provided all equipment with power and earthing connections, consideration must be given to the method of providing audio interconnection, and adequate screening of those interconnections. This must be done in a logical sequence to avoid problems, and assist in the localisation of problem equipment.

- a) Connect the Control Room Monitor system to the console, and check for any hum, buzz, or RFI. Only when you are satisfied with the quietness of the console and the monitor system should you proceed with the next step.
- b) Connect multitrack tape recorder, via noise reduction system if applicable, and again check that the system is still clean.
- c) Connect stereo tape recorders, studio monitors, echo and foldback sends one at a time, checking and isolating any connection which degrades performance.
- d) Connect all peripheral devices.
- e) Connect all microphone lines.

By following this sequence much time and future trouble will be saved, and the result will be a quiet, stable system.

3) Shielding

Audio equipment is supplied with a variety of input and output configurations, which must be taken into consideration when deciding where the screen connections should be made. There are three sources of unwanted signal being impressed on the screen, which are as follows:-

- i Extraneous electrostatic or electromagnetic fields.
- ii Noise and interference on the earth line.
- iii Capacitive coupling between the screen and signal wires.

To minimise the adverse affects of the unwanted coupling to the signal wires, it is important that the screen is connected at one end only, i.e. the screen must not carry any signal Any signal on the wires within the screen will be capacitively coupled to the screen, and this current will ultimately be returned to the source of the signal, either directly, if the screen is connected at the signal source end, or indirectly via the earthing system, if the signal is connected at the signal destination end. The indirect connection will cause an increase in high frequency cross-talk, and should be avoided wherever possible. Therefore, in general, always connect the shield only at the signal source end. high RF areas, the screen can also be connected to earth via a 0.01 micro Farad capacitor. This will present a short circuit at RF frequencies, thus lowering the effective shield impedance to ground. However, at low audio frequencies the reactance of the capacitor will be sufficiently high not to cause an earth loop problem.

Combinations of unbalanced, balanced and electronically balanced (differential) systems mean that there are nine interconnection permutations. The optimum of the screen in each case is shown in Table 1.

TABLE 1

	OUTPUT	INPUT	SCREEN
1	Unbalanced	Unbalanced	Source
2	Unbalanced	Balanced	Source
3	Unbalanced	Differential	Source
4	Balanced (Note 1) Balanced Balanced (Note 2)	Unbalanced	Destination
5		Balanced	Source
6		Differential	Destination
7	Differential (Note 3) Differential Differential	Unbalanced	Source
8		Balanced	Source
9		Differential	Source

- Note 1 The shield is connected to the destination earth point, which is opposite to normal practice, because the signal wires being shielded are referenced to the input earth, not the output earth.
- Note 2 If the output transformer is centre tapped to earth, the screen should be connected at the source.
- Note 3 When an active differential output is operated in unbalanced mode, it is very important that the output current returns to earth via the shortest, least reactive route. Check for instability at the output.

N.B.

- a) In all cases, use good quality twin screened audio cable. Check for instability at the output.
- b) Always connect both conductors at both ends, and ensure that the screen is only connected at one end.
- Do not disconnect the mains earth from each piece of equipment. This is needed to provide both safety and screen returns to the system star point.
- d) Equipment which has balanced inputs and outputs may need to be electrically isolated from the equipment rack and/or other equipment, to avoid earth loops.

It is important to remember that all equipment which is connected to the mains is a potential source of hum and interference, and may radiate both electrostatic or electromagnetic radiation. In addition, the mains will also act as a carrier for many forms of RF interference generated by electric motors, air-conditioning units, thyristor light dimmers etc. Unless the earth system is clean, all attempts to improve hum noise levels will be futile. In extreme cases there will be no alternative but to provide a completely separate and independant "technical earth" to replace the incoming "noisy earth". However, always consult your local electricity supply authority to ensure that safety regulations are not being infringed.

3.03 TS12 VARELCO CONNECTION EQUIVALENTS

TIE LINE	PHONES/AUX/EXT	2-TRACK/MONITOR
1	PHONES 1 L	STUDIO L
2	PHONES 1 R	STUDIO R
3	PHONES 2 L	· ALT L
4	PHONES 2 R	ALT R
5	GND	C/RM L
6	AUX 1	C/RM R
7	AUX 2	GND
8	AUX 3	2TA SEND L
9	AUX 4	2TA SEND R
10	AUX 5	2TB SEND L
11	AUX 6	2TB SEND R
12	GND	2TC SEND L
13	ECHO RET L	2TC SEND R
1 4	ECHO RET R	GND
15	GND	2TA RET L
16	EXT 1 L	2TA RET R
17	EXT 1 R	2TB RET L
18	EXT 2 L	2TB RET R
19	EXT 2 R	2TC RET L
20	GND	2TC RET R

EDACS

- 5 Edacs are supplied for 100 tie lines 3 Edacs are supplied for 24T machine (24 channel frame) 4 Edacs are supplied for 32T machine (32 channel frame)

- 1 Edac is supplied for Phones/Aux master
 1 Edac is supplied for Monitor/2T master

In the 24 channel frame, with ADC patchbay, only 1 Edac is supplied, for 20 tie-lines, as standard. An extra 3 Edacs can be fitted as an option.

4.00 OPERATION

4.01 INPUT/OUTPUT MODULE

CHANNEL

The Channel section with its rotary fader is usually the section that feeds an individual Mic or Line signal to the routing matrix.

The Channel input section is located near the top of the module, with its associated controls; Input Gain, phantom power on/off, mic/line select, pad and phase reverse.

The clearly defined "Equalisation" block, containing the 4-band semi-parametric EQ, the Hi Pass filter and the insert point can be located in either the channel path or the mix path, depending on the CH selection on individual modules. Both the HP Filter and the EQ can be switched in or out of the signal path independently.

The insert patch point is available in either the mix or channel path depending on CH selection. When the Insert point on the patch bay is positioned in the CHANNEL path this allows access to the Channel Signal Path immediately before the Channel Fader. (For instance, you may wish to compress a sound before it goes to tape, a compressor can be placed in the signal path via the channel insert send and return.)

When positioned in the MIX path the insert point allows access to the Mix signal path immediately before the Mix fader.

The Auxiliary sends, which are arranged in pairs, are usually fed from the Mix path but by pressing the CH button the Auxiliary sends 1 and 2 can be taken from the Channel path either Pre or Post Channel fader. This may be required, for example, when headphone mixes need to be derived from individual inputs, rather than from the Group output monitor, or when recording an instrument with a specific effect.

Pressing FLW on $\text{Aux}\ 3$ and 4 allows the signal to follow that in $\text{Aux}\ 1$ and 2, this can be either pre or post $\text{Aux}\ 1$ and 2 level depending on the internal jumper selection.

During RECORD the CHANNEL fader level would be set to about 0dB to allow the operator to increase the level by 10dB or to fade out completely, the Input Gain may be adjusted to give the required 0dB. (It may be necessary to switch in the 20dB PAD to avoid over-loading the input.) This will optimize the noise and headroom performance (dynamic range) of the signal path.

From the Channel fader the signal in the CHANNEL path may be assigned via the Pan Pot and routing matrix, to any of the Groups or the Main Mix. In addition, by selecting DIR the signal will be exclusively assigned to the corresponding Group Output.

Group Output level is controlled by the Group faders, located on the Group/FX return Module. When the SUB button is selected the group outputs are sent directly to the main stereo mix bus.

During remix, the Channel section need not lie idle, but may be used to provide extra inputs to the console or extra outputs; this facility may be individually selected on each module:-

- i. For use as an extra input (eg. for an FX return) the signal in the Channel Path may be routed to the Mix bus via the channel pan pot by pressing the MIX button on the routing matrix.
- ii. To use the Channel section on a module as an extra send, the button marked FDR disconnects the normal Channel input and feeds the Channel fader with a signal from the Mix fader below. Thus by using the Channel fader and pan pot it is possible to send to any or all of the 12 groups on the routing matrix. These can be used to provide extra mono or stereo sends during remix.

To prevent an accidental feedback loop, the operation of the FDR button prevents the Routing Matrix from being fed with the Tape send, ie. the Routing Matrix can only be fed with Tape Return when FDR is pressed.

The Channel path can be CUT independently of the Mix path by selecting CUT (located below the Channel fader). Also the Channel path has its own PFL (Pre-Fade Listen) facility. When PFL is selected a mono feed is sent to the main monitors, this facility is independent of the CUT button.

The MIX signal path normally goes to the stereo Mix bus. Input to the Mix path may be either Tape return or Tape send. When TPE is selected the input is the Tape return.

Should a tape have been recorded at a different level than the normal console working level of +4dBu then a tape trim is provided, this is located above the mix pan pot.

The Bounce (BNC) button disconnects the output from the pan pot from the mix Left and Right buses and sends the output to the routing matrix, OVERRIDING the normal Channel Input. This permits simple, instant track bouncing, or the formation of stereo audio sub-groups.

For example; a good balance of say 6 backing vocals has been achieved and the operator wishes to bounce this down into stereo, whilst maintaining the same balance.

- 1. Select BNC on the 6 modules which have been used for the individual vocals.
- 2. Select a pair of Groups (say 9 and 10) on the routing matrices of each of the 6 modules.
- 3. Adjust the Group faders (9,10) if necessary.

This has the overall effect of sending the balance being monitored directly to tape without adjustment, but INCLUDING any equalisation added.

A combination of BNC and FDR routes the output from the channel pan pot to the main mix bus and groups 9-12 and the output from the mix pan pot to groups 1-8, thus providing additional auxiliary sends during over-dubbing or during mix-down whilst maintaining the capability of audio sub-grouping. The input to the mix path can be either the tape send or the tape return. However, to prevent feed back you cannot route a group back into itself. The channel path can then feed the tape send directly if DIR is selected.

-					~ _ ~ _ ~ _ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~ ~	
	BNC	FDR	ch fdr I/P	ch pan O/P	mix fdr I/P	mix pan O/P
	0	0	channel	matrix	group/tape	mix
	1	0	channel	mix	group/tape	matrix
	0	1	mix(postfade)	matrix	group/tape	mix
	1	1	mix(postfade)	mix & 9-12	group/tape	1 – 8

0 - switch not pressed
1 - switch pressed
channel - signal in channel path
mix(postfade) - output from mix fader
matrix - routing matrix (buses 1-12)
mix - main mix bus

The Mix path SOLO can be either PFL or SIP (solo-in-place) depending on selection on the Monitor master. If SIP (Solo in place) is selected then the soloed signal will appear in its normal stereo position. If PFL is selected then a mono feed will be fed to the control room monitors.

The programmable electronic muting is extremely useful when a number of signals are required to be cut at the same time. A mute may be programmed by selecting either A or B on the I/O module. Pressing Master Mute A or B on the Monitor master will mute all the I/O modules with that particular selection. (Master Mute A cuts all those channels with A selected and Master Mute B cuts those with B selected.) Both A and B may be selected together on any module.

The Peak Led indicates overload in BOTH signal paths (Channel and Mix) in the module at the following points:Post Channel input amplifier, Post insert return and Post equaliser. When any of the points exceeds +15dBu (6dB below clipping) the LED will light. Even short duration transients will register since reaction time is very short, but the decay is purposel slow. Both positive and negative peaks are detected.

Metering for each tape track is provided by a mechanical VU meter in the overbridge. LED metering is available as an option, as are PPMs for broadcast specification.

The Meter above each module always relates to the Mix path signal. When the Mix path on the module is switched to Group, then the meter will read the Tape send; when the Mix path is switched to Tape then the meter will monitor the tape return.

4.02 MASTER MODULE

HEADPHONE CIRCUITS

A headphone mix may be built up by using Aux 1 and 2 sends and the studio monitor feed. The two headphone circuits allow for two different headphone mixes. For instance; you have a stereo cue mix being sent via the Studio monitor mix to headphone circuit 1. One of the artists requests more of himself.

- 1 Tell him to plug into headphone circuit no 2
- 2 Select STUDIO and press say AUX 1 on the select buttons on Phones 2
- 3 Turn up Auxiliary send no 1 on the appropriate input module to suit.

The artist has the same mix as before, but with the addition of more of himself.

MONITORING FACILITIES

Outputs are provided for three sets of monitor speakers, these would normally be the main Control Room monitors, outputs for an Alternative pair of monitor speakers and Studio monitor speakers. Both the studio and control room monitors are provided with a variable gain control.

By pressing the ALT button the control room output is routed to the alternative pair of control room monitors.

The Control Room monitors are provided with a DIM control to allow the output to be attenuated by 20dB. Pressing DIM introduces the attenuator into the signal path. This is useful for checking balance during Mix down. Also, conversation and telephone calls etc in the Control Room can be heard without having to compete with the band!

The MONO button sums the left and right outputs for a mono compatability check, as well as being useful for checking for out of phase microphones, pick-ups, etc.

The source for the monitors can be mix left and right or one of five external sources, (2-tracks A, B, or C, Ext 1 or 2). The studio monitor source select is separate from the Control Room monitor select.

SOLO SELECT

Housed in the same section as the monitor select switches is the SIP button. Normally when channel solo is selected the signal monitored is the PFL signal. When SIP is selected the signal is heard in its stereo position on the main mix outputs.

TALKBACK FACILITIES

The Talkback signal can be routed to the Tape sends, Headphone circuits and the Studio monitors by selecting the appropriate button.

MUTE SELECT

The programmable electronic muting system is a noiseless design with defined attack and decay times, this allows the operator to perform fast mutes without fear of 'clicks' occuring.

The Master Mute controls are located above the main stereo mix fader. A mute can be programmed by selecting either A or B on individual modules. Pressing Master Mute A on the Monitor master will cut those Mix paths with Mute A primed. Mute B operates in an identical fashion.

Programmable mutes can be very convenient when a number of signals need to be cut at the same time. On the TS12 console the electronic muting allows a mute to be primed prior to when the cut is to take place and you need only press one button when the mute is to occur, no more running out of hands!

MAIN FADERS

The Mix stereo fader is a long throw, high quality, linear fader with +10dB at the top of its travel.

METERING

The Mix outputs are provided with wide scale mechanical VU meters for both left and right outputs. The Phase correlation meter displays the phase relationship between the left and right outputs giving the operator a visual indication that an in-phase signal is present at the main mix output. By selecting tape returns on the Control Room Monitor switches the operator can also check that no phase errors have been introduced between the console and the tape machine, and that there should be no problems during disc cutting.

In-phase signals are indicated by the illuminated segment moving to the right-hand side of the display. For out-of-phase signals the segment will move to the left. For signals which do not have a clearly defined phase relationship the segment could move to an indeterminant position.

4.03 Auxiliary master and Echo return

OSCILLATOR SELECT

Before any recording session commences it is essential that the tape machines, both multitrack and 2-track master are correctly aligned.

The internal oscillator on the TS12 may be routed to the 12 Group Outputs and hence the multitrack machine by pressing the button marked TAPE on the Oscillator section. The alignment of the tape machine may be checked at any frequency between 20Hz and 20kHz.

The output of the oscillator can also be routed to other parts of the console using the oscillator output on the patch bay, this is useful for fault finding purposes.

AUX MASTERS

There are 6 auxiliary send master level controls. Each master has its own AFL (After-Fade Listen) facility. The LED illuminates when AFL is pressed.

When AFL is selected the signal currently on the control room monitors is replaced by the AFL signal.

ECHO RETURN

The stereo echo return is used to bring stereo effects back in to the console. The stereo signal can then be routed to any pair of the six auxiliaries, any pair of groups 1-12 and/or the main mix bus.

4.04 OPTIONAL EFFECTS RETURN MODULE

The optional Effects Return module further enhances the versatility of the Soundcraft TS12 in-line console. The Effects Return module provides 4 additional inputs to the console which feed the main mix bus. The signal may also be sent to Aux 1 and 2.

Input to the module is via the associated patchcard. By using molex connectors, located at the bottom of the FX return patchcard, equipment which is often used can be permanently patched into the console. Alternatively, input to the module can be via the associated input jack socket on the patchcard. The FX return has balanced inputs.

5.00 MAINTENANCE

Every console that leaves Soundcraft undergoes a thorough testing at all stages of manufacture. These tests include individual testing of every function on all the PCB's, a thorough testing of all the functions of the completed mixer, a soak test of 48 hours before the final test, which consists of listening, measuring and mechanical function checks prior to packaging and shipment. In this way we try to ensure that any faulty components or manufacture show up long before the console leaves the company. Thus a long and trouble-free life can be expected.

Although all Soundcraft Consoles have been designed with long term reliability in mind, it is inevitable that occasional maintenance will be required. However, due to the amount of attention given to the problems of maintenance during the design stages of this console, and the modular construction, servicing tends to be extremely simple to carry out, with the minimum of test equipment needed to isolate and rectify faults.

5.01 General Fault Finding

All signal electronics are configured around high slew rate, low noise integrated circuits. The microphone amplifier is a proprietory design, utilizing discrete input transistors, a noise cancelling front end, differentially summed via a low noise integrated circuit, using an Active Feedback Network to yield a very low noise design.

The use of integrated circuits means that the majority of audio faults can be repaired by simply replacing the I.C., having first isolated the fault to a particular stage in the signal chain. The isolation can often be done without even having to remove the module from the console, by judicious use of insert points, and/or switching the module to various modes. As with all servicing a good knowledge of the basic signal flow is necessary for best results. Each module should be viewed as a number of signal blocks, through which the signal must flow. the signal appears at the input to a block, but not at the output, then the fault usually lies within that block. dividing a module into individual sections, what at first appears to be an extremely complicated piece of equipment can be simplified into a series of sequential stages. This is the basic step in all types of fault finding, and usually requires no more than a certain amount of logical thought. Servicing a console is more a matter of clear thinking and having an understanding of what should be happening, than having a highly developed technical knowledge.

To illustrate the method of logical fault finding, let us assume that we have a non functioning channel input, in both microphone and LINE modes.

The first step is to ensure that a fault really does exist! Check that the module is in the correct mode of operation, and that no effects are inserted in the insert points, which may be interrupting the signal flow.

If in doubt about the module operation, set up an adjacent module in exactly the same way, which will allow a direct comparison between a working and possible non-working module.

Route the channel directly to the Mix bus by selecting MIX on the routing section and ensuring that the console status is MIX. Using the internal oscillator set to 1kHz, select P/BAY and patch the oscillator signal into the channel input at the patchbay. NB: Ensure that the +48V phantom power is off. If all is well, an undistorted signal should now be heard when the fader is pushed up. More likely, because of the fault it won't.

Large sections of the module circuitry can be by-passed by switching out the Hi-pass filter and the equalizer.

If switching out a section causes the signal to re-appear, then the fault is located in that section, which can then be traced at component level, by removing the module from the console frame, and reconnecting it via extender cables provided.

With the module installed on extender cables, access is now available to parts of the module, and the signal may be traced through the various stages, using an oscilloscope, millivoltmeter, or even high impedance headphones. Refer to the Block Schematic which shows the signal flow through the modules. When a point is reached where the signal is not present, or is distorted, the probable faulty components can be checked out and if necessary replaced. Integrated cicuits, due to their internal complexity, are the most likely cause of problems, followed by mechanical components such as switches and faders, which are susceptible to physical contamination from oxidisation, dust and liquids.

5.02 Removing Modules

Remove the 2 module retaining screws, located at the top and bottom of the module, and carefully remove the module.

The module will still have some cables attached, these will be the 10-way IDC and the two methode connectors located at the top of the module. The cables attached should be long enough to remove the module.

An extender PCB can now be plugged between main 40-way IDC on the mother loom and the module.

Although damage to the module will not be caused by plugging and unplugging the module with the power still switched on, this is not recommended for the inexperienced, as it is possible to bend the connector pins if care is not taken.

When replacing the module the 40-way connector should re-locate into the mother loom.

5.03 Removing Patch Cards

The entire patchbay is made up from rows of plug in patch cards, constructed with printed circuit mounting jack sockets. Each row can be easily removed for cleaning and replacement if this should prove necessary.

The patchbay is hinged to allow easy access. To lift up the patchbay undo the retaining screws located at the top of the patchbay. This will allow the whole assembly to be lifted up and tilted forward side. Take care to avoid scratching the paint.

Having lifted up the patchbay locate the card which is to be removed and unplug the ribbon cables connected to it, and, if necessary, the adjacent cards. Remove the two screws holding the card to the panel and withdraw the card from the assembly.

Reassembly of the patchbay is the reverse of the above procedure.

5.04 VU Meter Alignment

The master VU meters are driven from the PCB located below the meters, this also contains the phase correlation meter. The potentiometers to adjust these meters are located on the meter drive PCB.

OVU is normally adjusted to indicate a line level of +4dBu, ie. a level of 1.228volts. However, it can be readjusted to indicate a different line level if required by the preset potentiometer on the drive card.

Connect a millivoltmeter to the output of the console. Route the oscillator set to 1kHz to the Mix bus by selecting TAPE on the oscillator section of the Master Module. Adjust the Mix output level to read the required level on the millivoltmeter. (Normally this would be +4dBu.) Adjust the VU drive preset to indicate OVU on the VU meters.

5.05 Lamp Replacement

To replace the lamps first remove the cover on the back of the meter bridge by removing the retaining screws located at the top of the cover. The lamps are located on top of the meter and can be taken out by removing the backplate which holds the backplate of the bulb holder to the meter bridge and desoldering the bulbs. New bulbs can now be soldered in place.

6.00 SOUNDCRAFT RECOMMENDED WARRANTY

(This warranty applies to sales within the UK and should form the basis of the warranty offered by the overseas vendor of Soundcraft products.)

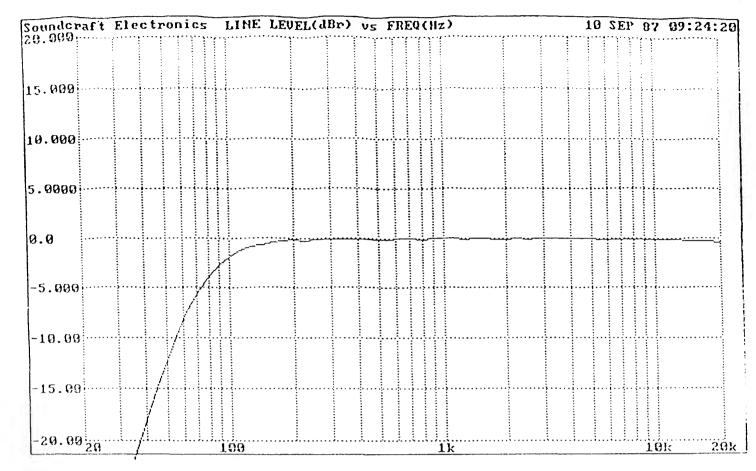
- 1. `Soundcraft' means Soundcraft Electronics Ltd.

 End User' means the person who first puts the equipment into regular operation.

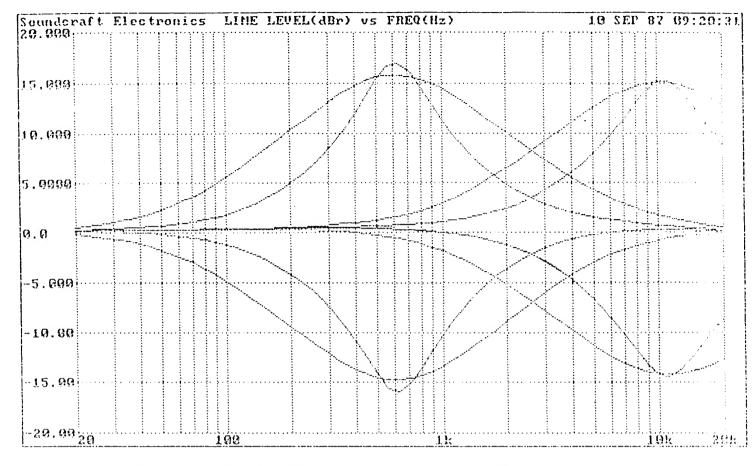
 Dealer' means the person other than Soundcraft (if any) from whom the End User purchased the Equipment, provided such a person is authorised for this purpose by Soundcraft or its accredited Distributor.

 Equipment' means the equipment supplied with this manual.
- 2. If within the period of twelve months from the date of delivery of the Equipment to the End User it shall prove defective by reason only of faulty materials and/or workmanship, (but not faulty design), to such an extent that the effectiveness and/or usability thereof is materially affected the Equipment or the defective component should be returned to the Dealer or to Soundcraft and subject to the following conditions the Dealer or Soundcraft will repair or at its option replace the defective components. Any components replaced will become the property of Soundcraft.
- 3. Any Equipment or component returned will be at the risk of the End User whilst in transit, (both to and from the Dealer or Soundcraft), and postage must be prepaid.
- 4. This warranty shall only be available if:
 - a) the Equipment has been properly installed in accordance with instructions contained in Soundcraft's manual; and
 - b) the End User has notified Soundcraft or the Dealer within 14 days of the defect appearing; and
 - no persons other than authorised representatives of Soundcraft or the Dealer have effected any replacement of parts maintenance adjustments or repairs to the Equipment; and
 - d) the End User has used the Equipment only for such purposes as Soundcraft recommends, with only such operating supplies as meet Soundcraft's specifications and otherwise in all respects in accordance with Soundcraft's recommendations.

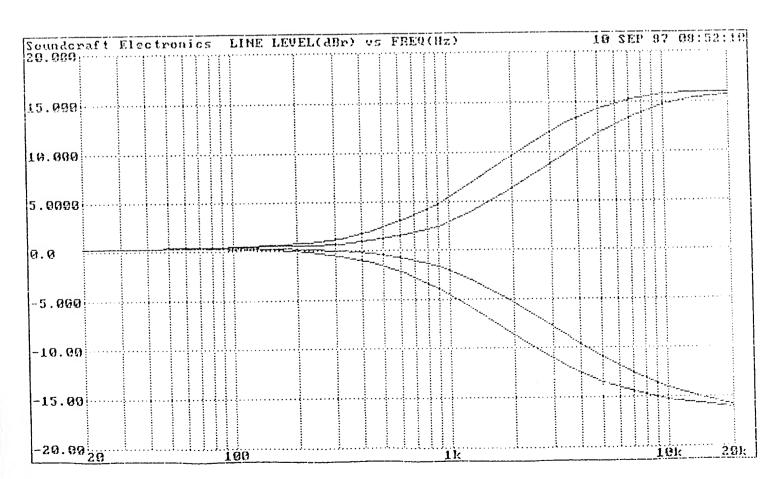
- 5. Defects arising as a result of the following are not covered by this Warranty: faulty or negligent handling, chemical or electro-chemical or electrical influences, accidental damage, Acts of God, neglect, deficiency in electrical power, air-conditioning or humidity control.
- 6. The benefit of this Warranty may not be assigned by the End User.
- 7. End Users who are consumers should note their rights under this Warranty are in addition to and do not affect any other rights which they may be entitled against the seller of the Equipment.



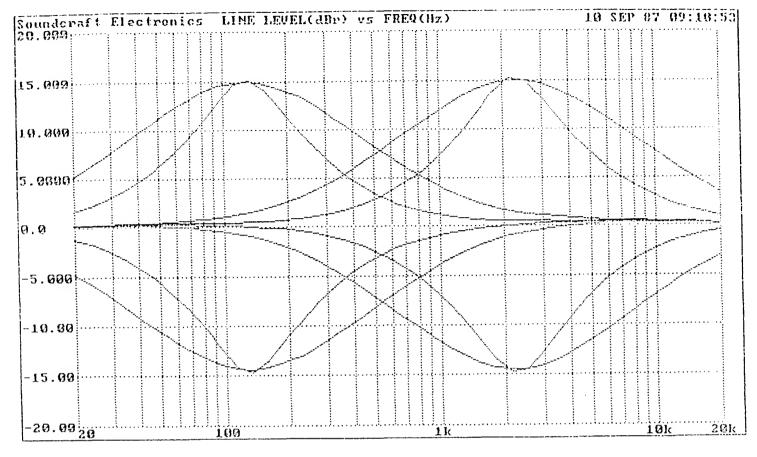
TS12 I/O MODULE EQUALISER- LOW CUT FILTER.



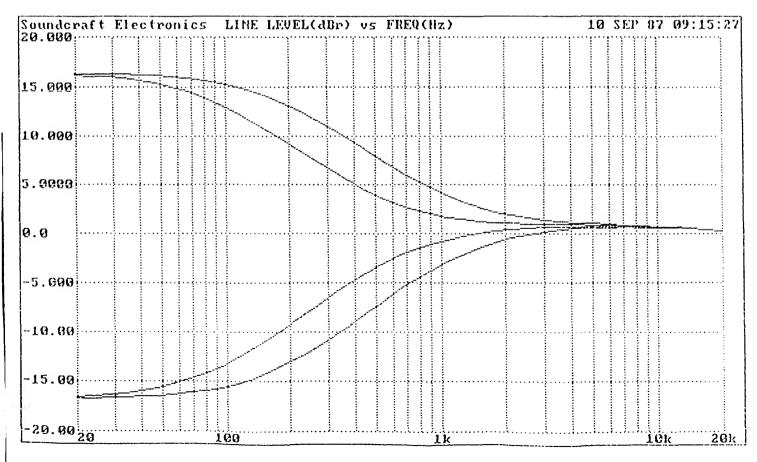
TS12 I/O MODULE EQUALISER- HI MID FILTER.



TS12 I/O MODULE EQUALISER- HE FILTER.



TS12 I/O MODULE EQUALISER- LO MID FILTER.



TS12 I/O MODULE EQUALISER- LE FILTER.

TS.12 MOTHERLOOM 40 Way IDC Buss Connector

```
Master Mute 'B'
1
    Master Mute 'A'
2
    PFL Enable
3
4
    SIP Mute
5
   +24V
    +24V
6
7
    24V GND
    24V GND
8
9
    PFL Audio
10
    Aux Summing (Dustbin) GND
    Group Bus 1
11
12
    Group Bus 2
    Group Bus 3
13
14
    Group Bus 4
15
    Group Bus 5
    Group Bus 6
16
    Group Bus 7
17
    Group Bus 8
18
19
    Group Bus 9
20
    Group Bus 10
21
    Group Bus 11
22
    Group Bus 12
23
    GND
24
    GND
25
    MIX Bus Left
    MIX Bus Right
26
27
    GND
28
    GND
29
    Aux Bus 6
30
   Aux Bus 5
31
    Aux Bus 4
32
    Aux Bus 3
    Aux Bus 2
33
34
    Aux Bus 1
35
    Chassis GND
36
    +48V
37
    +17V
38
   +17V
39
    -17V
40
    -17V
```

I/O Right Hand

CN.1 20 Way IDC Connector

- 1 LED Chain From TR9
- 2 Output From Hi-pass Filter
- 3 Input To Hi-pass Filter
- 4 EQ Output/Insert Send Buffer
- 5 Insert Send
- 6 EQ Input
- 7 Mix Pre/Chan Pre via SW10 'CH'
- 8 Aux 1&2 Feed
- 9 Mix Post
- 10 Aux 3&4 Feed
- 11 Mix Pre/Post via SW12 'PRE'
- 12 Aux 5&6 Feed
- 13 Input Pre-amp Output From 'PFL' & 'CH' Switch
- 14 Input To 'CH' Fader
- 15 Normal Group Output Buffer
- 16 Chan Post Feed
- 17 -17V
- 18 -17V
- 19 +17V
- 20 +17V

I/O Right Hand

CN.2 20 Way IDC Connector

- 1 Mix Bus Right To CN3 Pin26
- 2 Mix Bus Left To CN3 Pin25
- 3 GND
- 4 GND
- 5 GND
- 6 GND
- 7 Group 9 & 11 Routing Feed
- 8 Group 10 & 12 Routing Feed
- 9 Chan Pan
- 10 Chan Pan
- 11 Group 1,3,5 & 7 Routing Feed
- 12 Group 2,4,6 & 8 Routing Feed
- 13 Mix Pan
- 14 Mix Pan
- 15 Mix Post Feed To 'FDR'
- 16 Tape Send EBOS Feed
- 17 Tape Return Buffer Output
- 18 Meter Feed & Input to Mix 'TRIM' Buffer
- 19 CUT Line From Automated L/H I/O
- 20 LED Chain To LED 1 'CH'

I/O Right Hand Connectors

CN.4 3 Way Microphone Input

- 1 GND
- 2 Mic. Input -ve
- Mic. Input +ve

CN.5 16 Way IDC Connector

- 1 Normal Group -ve
- Normal Group +ve
- 3 Tape Return -ve
- 4 Tape Return +ve
- 5
- GND Tape Send EBOS GND Tape Send EBOS Tape Send -ve 6
- 7
- 8 Tape Send +ve
- Insert Return -ve 9
- 10 Insert Return +ve
- 11 Insert Send
- 12 GND Insert Send
- 13 GND
- 14 GND
- 15 Line Input -ve
- 16 Line Input +ve

CN.6 2 Way Meter Feed

- 1 GND
- Meter Feed

CN.8 3 Way Auto Mute Interface

- 1 -ve Logic Rail (derived)
- CUT Line Return
- CUT Line Send 3

Mix Monitor Master CN.1 16 Way IDC Connector

- 1 EXT 1 Right -ve
- EXT 1 Right +ve 2
- 3 EXT 1 Left -ve
- EXT 1 Left +ve
- Phones In Right -ve
- Phones In Right +ve
- 7 Phones In Left -ve
- Phones In Left +ve 8
- Phones 2 Right GND CMP
- 10 Phones 2 Right +ve
- 11 Phones 2 Left GND CMP
- 12 Phones 2 Left +ve
- 13 Phones 1 Right GND CMP
- 14 Phones 1 Right +ve
- 15 Phones 1 Right GNP CMP
- 16 Phones 1 Right +ve

CN.2 16 Way IDC Connector

- 2Tk C Right -ve
- 2Tk C Right +ve 2
- 2Tk C Left -ve
- 2Tk C Left +ve
- 2Tk B Right -ve
- 2Tk B Right +ve 6
- 7 2Tk B Left -ve
- 8 2Tk B Left +ve
- 2Tk A Right -ve
- 10 2Tk A Right +ve
- 2Tk A Left -ve 11
- 2Tk A Left +ve 12
- 13 EXT 2 Right -ve
- EXT 2 Right +ve 14
- 15 EXT 2 Right -ve
- EXT 2 Right +ve 16

Mix Monitor Master CN.3 16 Way IDC Connector

- Studio Monitor Select Right -ve
- 2 Studio Monitor Select Right +ve
- 3 Studio Monitor Select Left -ve
- 4 Studio Monitor Select Left +ve
- 5 Select Switch GND
- 6 Select Switch GND
- 7 MIX Right -ve
- 8 MIX Right +ve
- 9 MIX Left -ve
- 10 MIX Left +ve
- 11 Select Switch GND
- 12 Select Switch GND
- 13 EXT 1 Right -ve
- 14 EXT 1 Right +ve
- 15 EXT 1 Left -ve
- 16 EXT 1 Left +ve

CN.4 16 Way IDC Connector

- 1 C/RM Right GND CMP
- 2 C/RM Right +ve
- 3 C/RM Left GND CMP
- 4 C/RM Left +ve
- 5 GND
- 6 GND
- 7 ALT Right GND CMP
- 8 ALT Right +ve
- 9 ALT Left GND CMP
- 10 ALT Left +ve
- 11 GND
- 12 GND
- 13 Studio Right GND CMP
- 14 Studio Right +ve
- 15 Studio Left GND CMP
- 16 Studio Left +ve

Mix Monitor Master CN.5 16 Way IDC Connector

- 1 MIX Right -ve
- 2 MIX Right +ve
- 3 MIX Right EBOS GND
- 4 MIX Right EBOS GND
- 5 MIX Left -ve
- 6 MIX Left +ve
- 7 MIX Left EBOS GND
- 8 MIX Left EBOS GND
- 9 MIX Insert Return Right
- 10 MIX Insert Return Right
- 11 MIX Insert Send Right GND CMP
- 12 MIX Insert Send Right +ve
- 13 MIX Insert Return Left
- 14 MIX Insert Return Left
- 15 MIX Insert Send Left GND CMP
- 16 MIX Insert Send Left +ve

CN.7 10 Way IDC Connector

- 1 +17V
- 2 +17V
- 3 Talk to Phones
- 4 N/C
- 5 Talk to Studio
- 6 N/C
- 7 Talk to Tape
- 8 GND
- 9 Talkback Audio
- 10 GND

CN.8 & CN.9 3 Way Fader Connectors

- 1 Fader Send
- 2 GND
- 3 Fader Return

CN.10 3 Way Connector

- 1 PFL Audio Out
- 2 GND
- 3 Talkback Audio Out

CN.11 3 Way Connector

- 1 GND
- 2 Talkback Microphone Output
- 3 Talkback Microphone Power

Group/Fx Module

CN.1 10 Way IDC Connector

- 1 Fx Return Input Upper -ve
- 2 Fx Return Input Upper +ve
- 3 Group Output EBOS GND Left
- 4 Group Output EBOS GND Left
- 5 Group Output -ve
- 6 Group Output +ve

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- 7 Group Insert Return -ve
- 8 Group Insert Return +ve

Group Insert Send +ve

- 9 Group Insert Send GND

CN.2 10 Way IDC Connector

- 1 Fx Return Input Lower -ve
- 2 Fx Return Input Lower +ve
- 3 Group Output EBOS GND Left
- 4 Group Output EBOS GND Left
- 5 Group Output -ve
- 6 Group Output +ve
- 7 Group Insert Return -ve
- 8 Group Insert Return +ve
- 9 Group Insert Send GND
- 10 Group Insert Send +ve

CN.4 & CN.5 Group Fader Connectors

CN.4 Left Fader CN.5 Right Fader

- 1 Fader Send
- 2 GND
- 3 Fader Return

CN.6 & CN.7 Fx Return Fader Connectors

CN.6 Fx Left CN.7 Fx Right Stereo Fader

- 1 Send
- 2 GND
- 3 Return

Aux/Echo

CN.1 16 Way IDC Connector

- 1 AUX 3 EBOS GND
- 2 AUX 3 EBOS GND
- 3 AUX 3 Output -ve
- 4 AUX 3 Output +ve
- 5 AUX 2 EBOS GND
- 6 AUX 2 EBOS GND
- 7 AUX 2 Output -ve
- 8 AUX 2 Output +ve
- 9 AUX 1 EBOS GND
- 10 AUX 1 EBOS GND
- 11 AUX 1 Output -ve
- 12 AUX 1 Output +ve
- 13 OSC. EBOS GND
- 14 OSC. EBOS GND
- 15 OSC. Output -ve
- 16 OSC. Output +ve

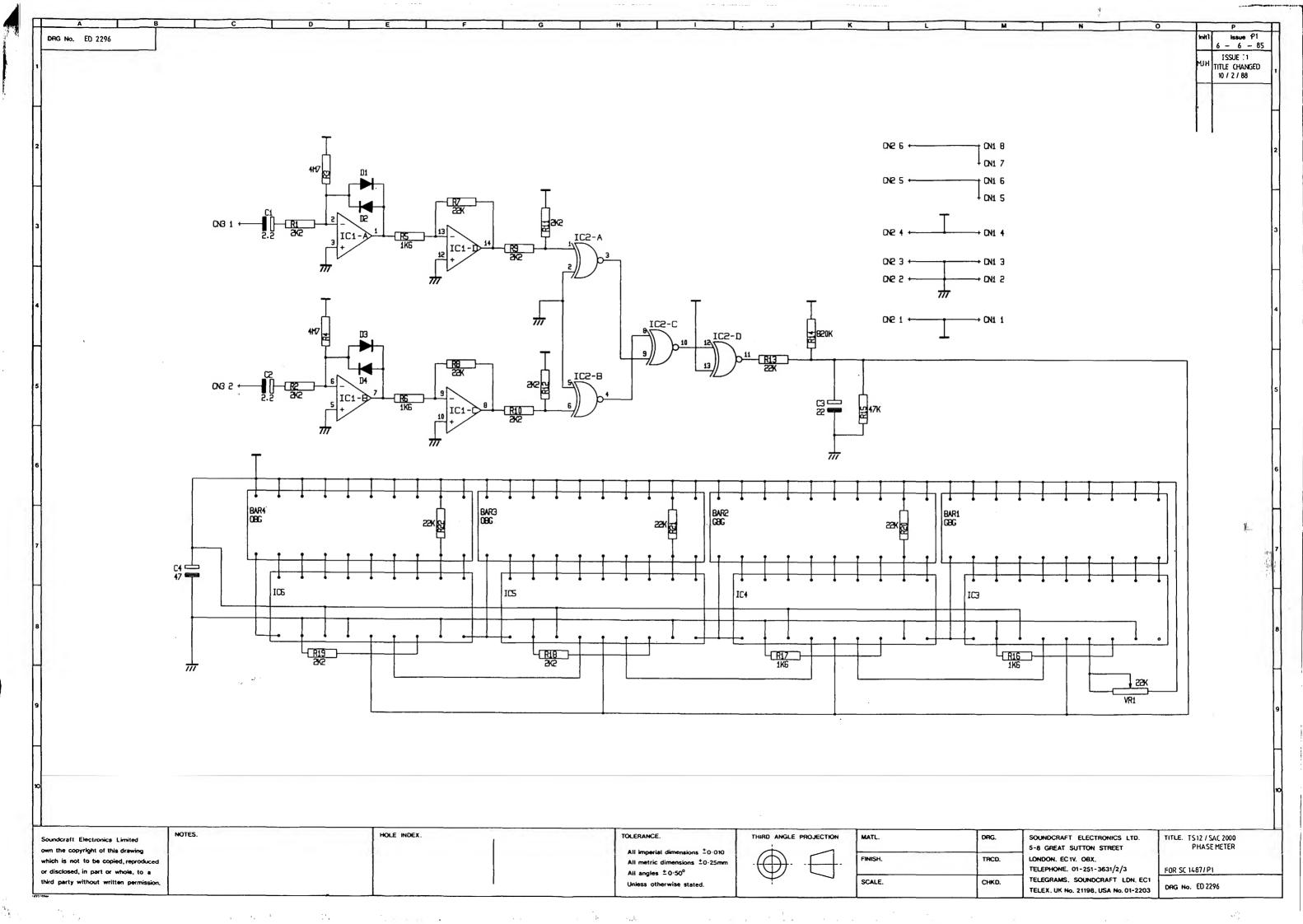
CN.2 16 Way IDC Connector

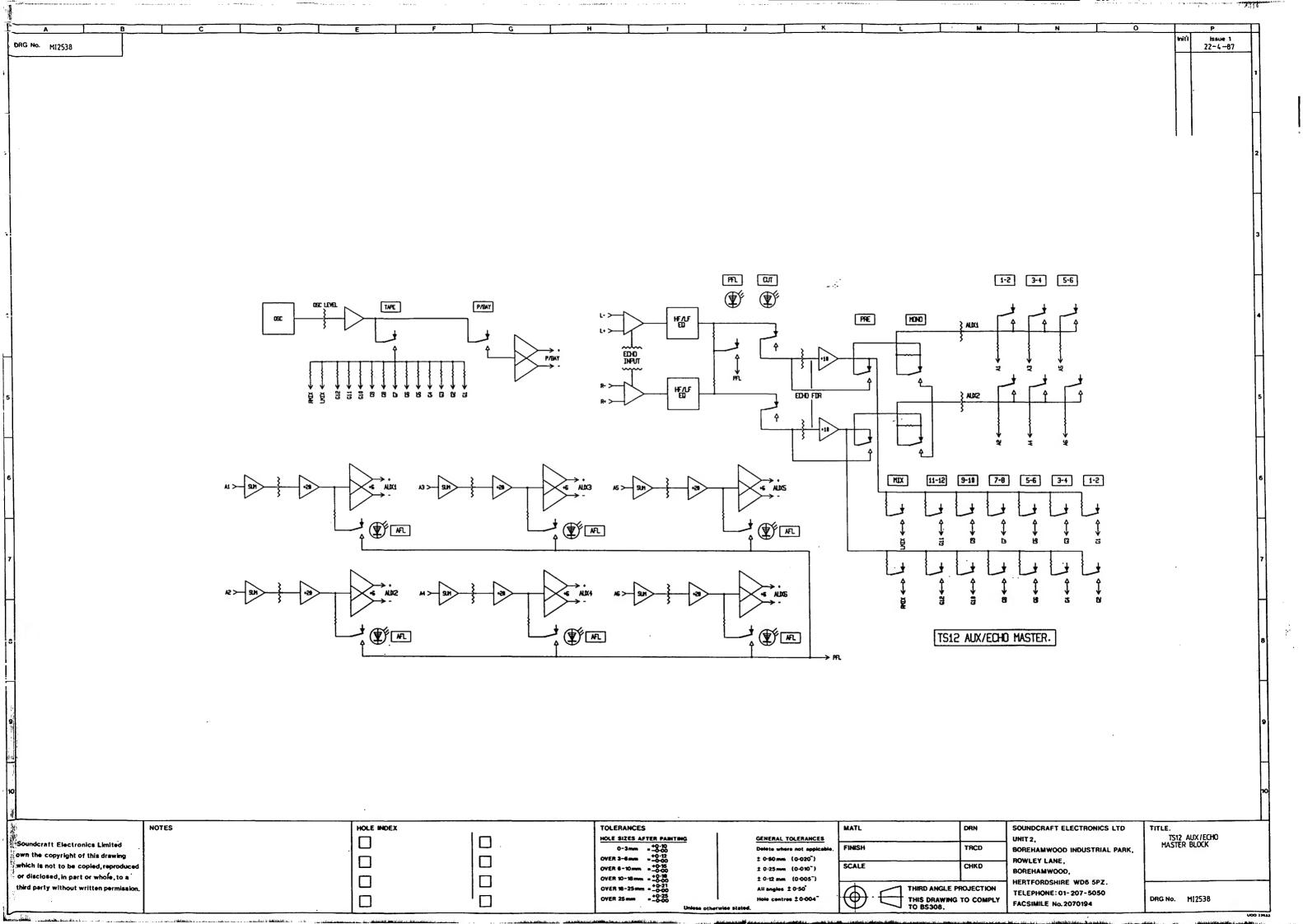
- 1 Echo Return Right -ve
- 2 Echo Return Right +ve
- 3 Echo Return Left -ve
- 4 Echo Return Left +ve
- 5 AUX 6 EBOS GND
- 6 AUX 6 EBOS GND
- 7 AUX 6 Output -ve
- 8 Aux 6 Output +ve
- 9 AUX 5 EBOS GND
- 10 AUX 5 EBOS GND
- 11 AUX 5 Output -ve
- 12 Aux 5 Output +ve
- 13 AUX 4 EBOS GND
- 14 AUX 4 EBOS GND
- 15 AUX 4 Output -ve
- 16 AUX 4 Output +ve

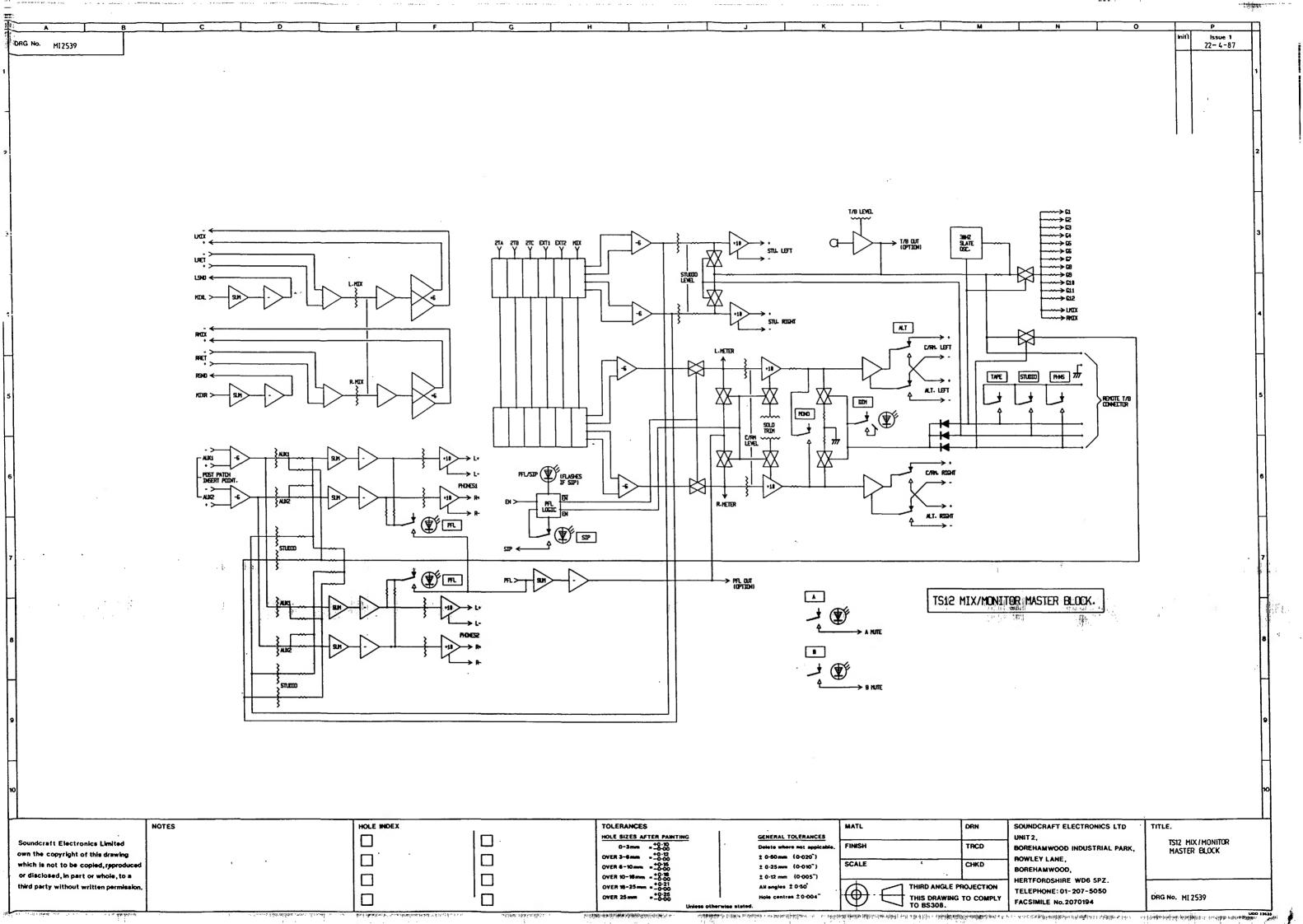
Aux/Echo

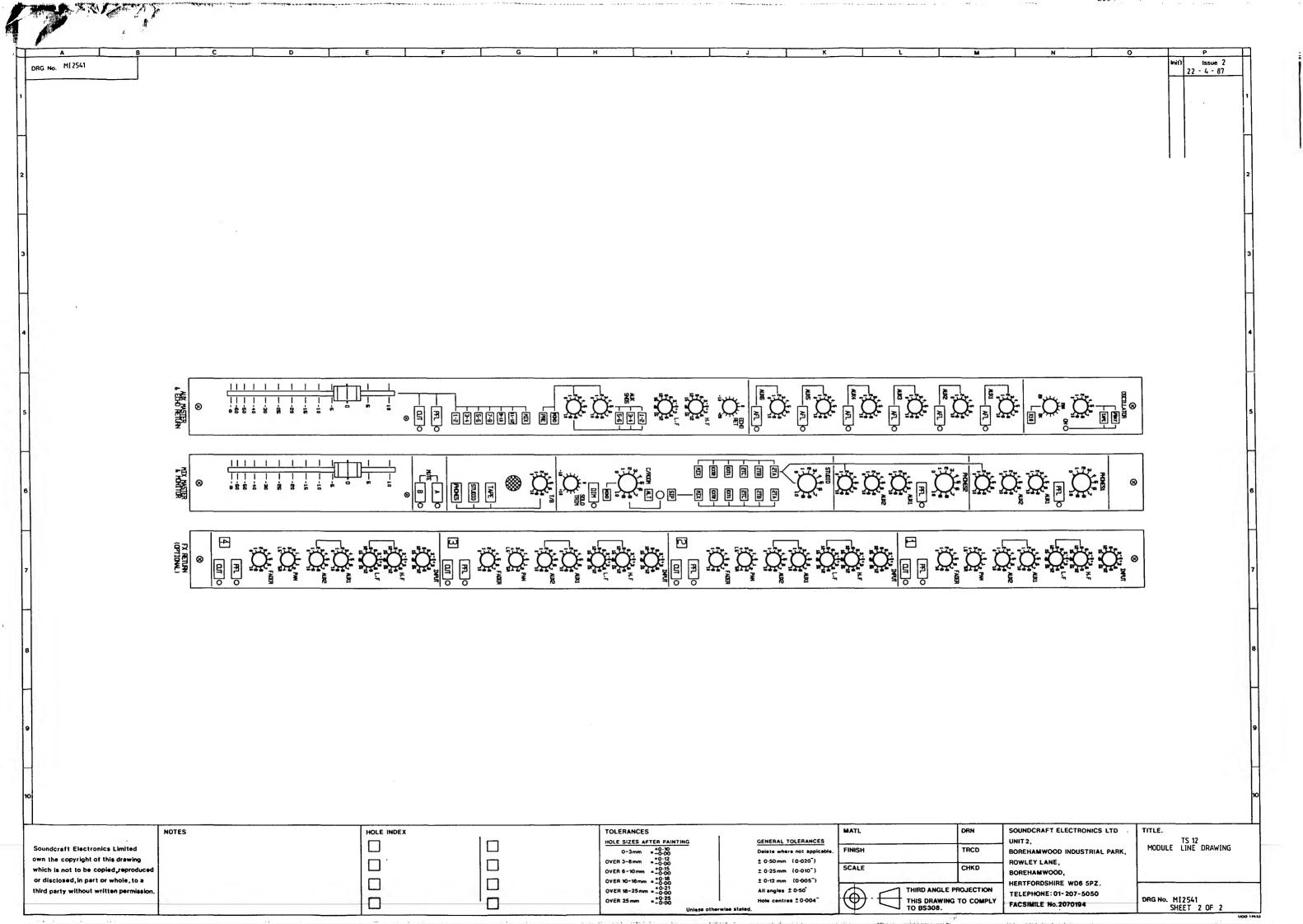
CN.4 & CN.5 3 Way Fader Connectors

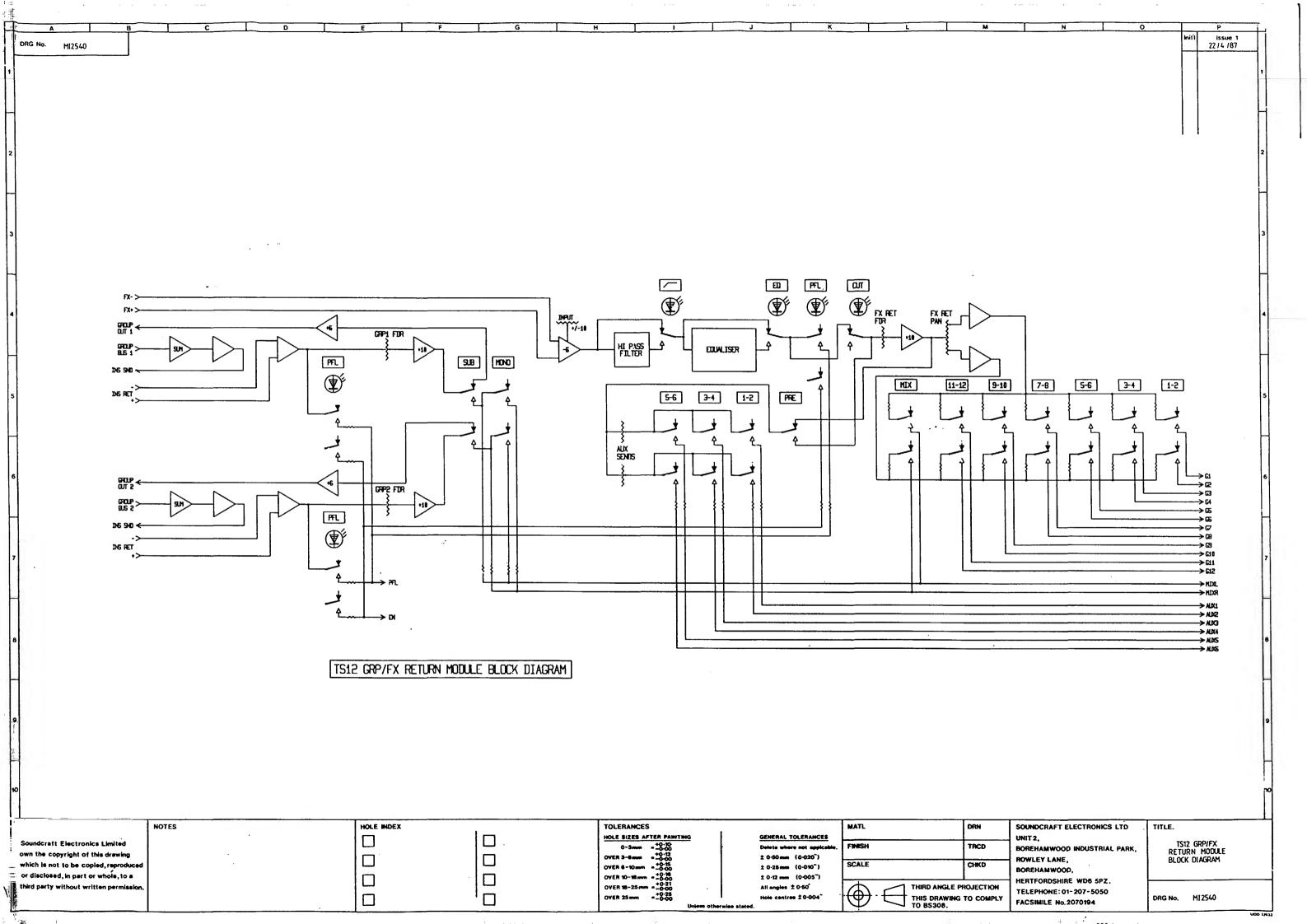
- 1 Fader Send
- 2 GND
- 3 Fader Return

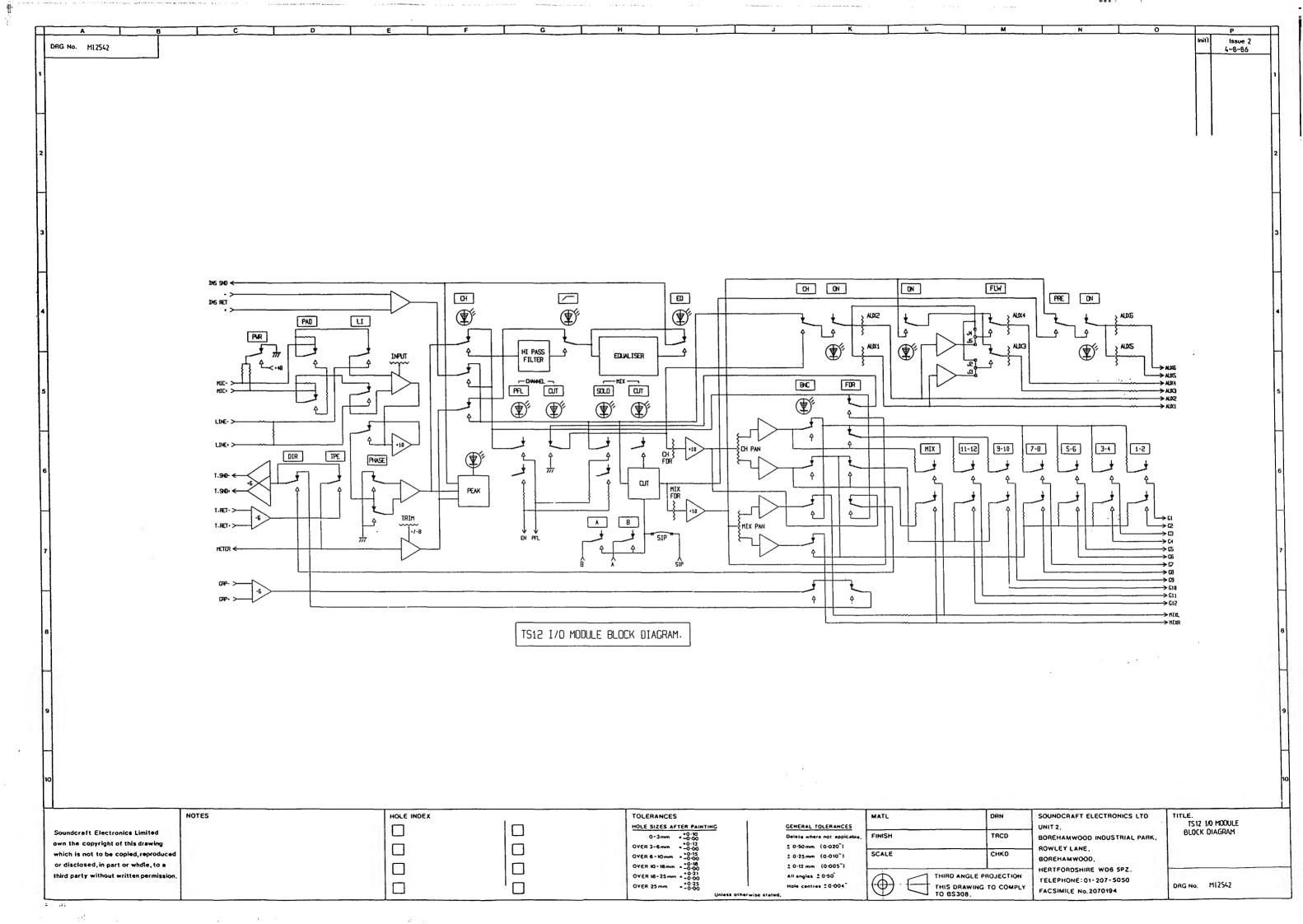


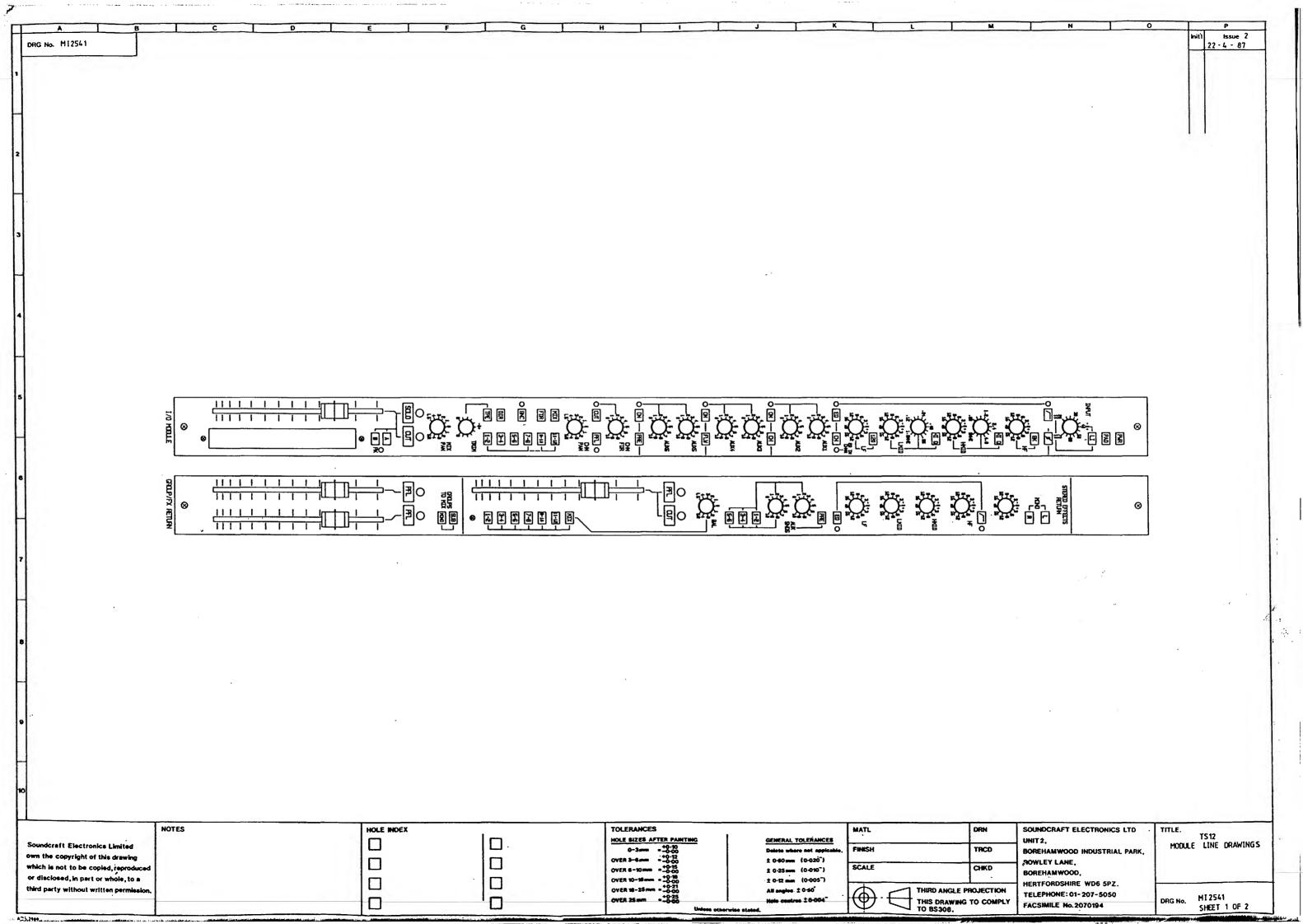


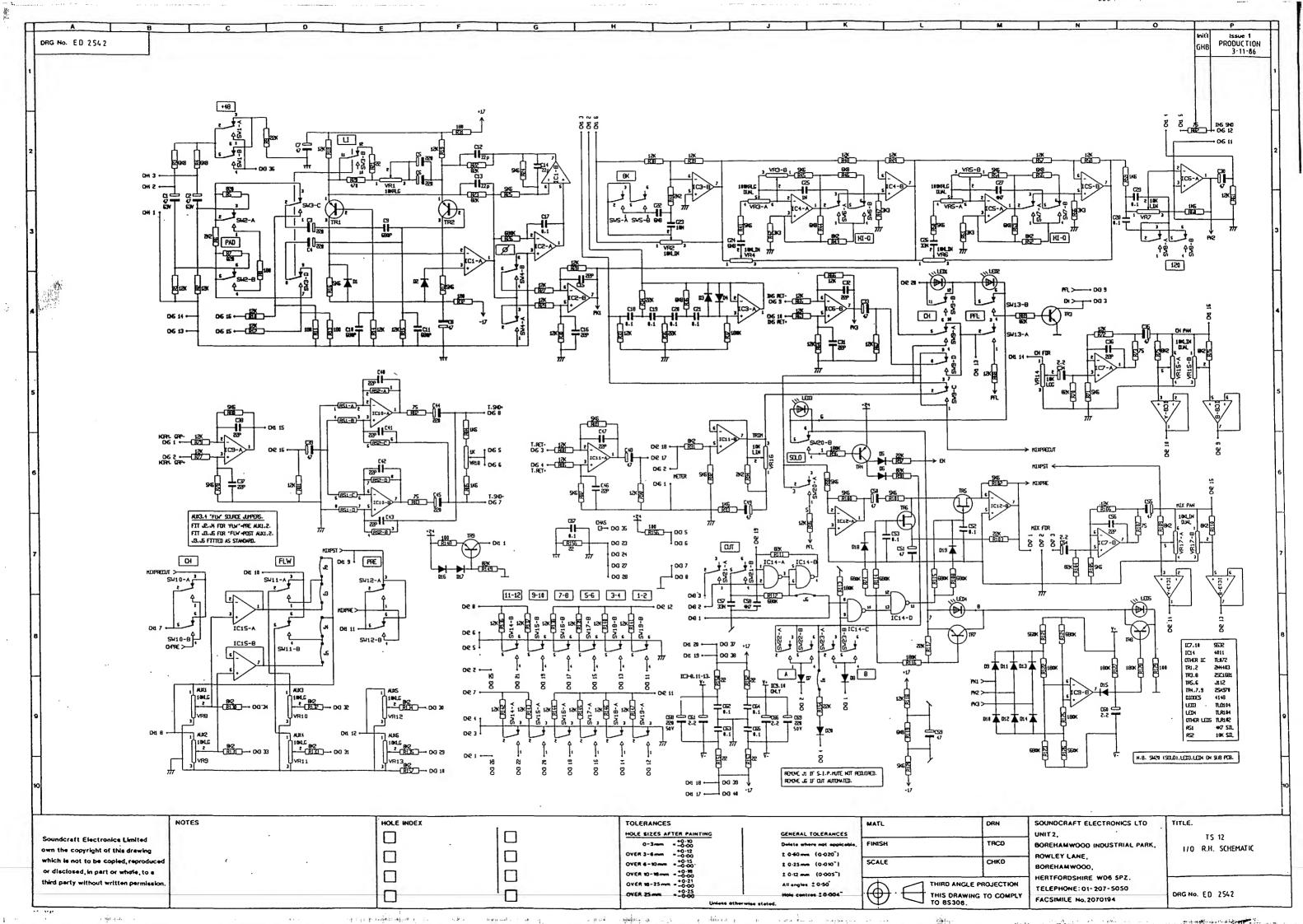


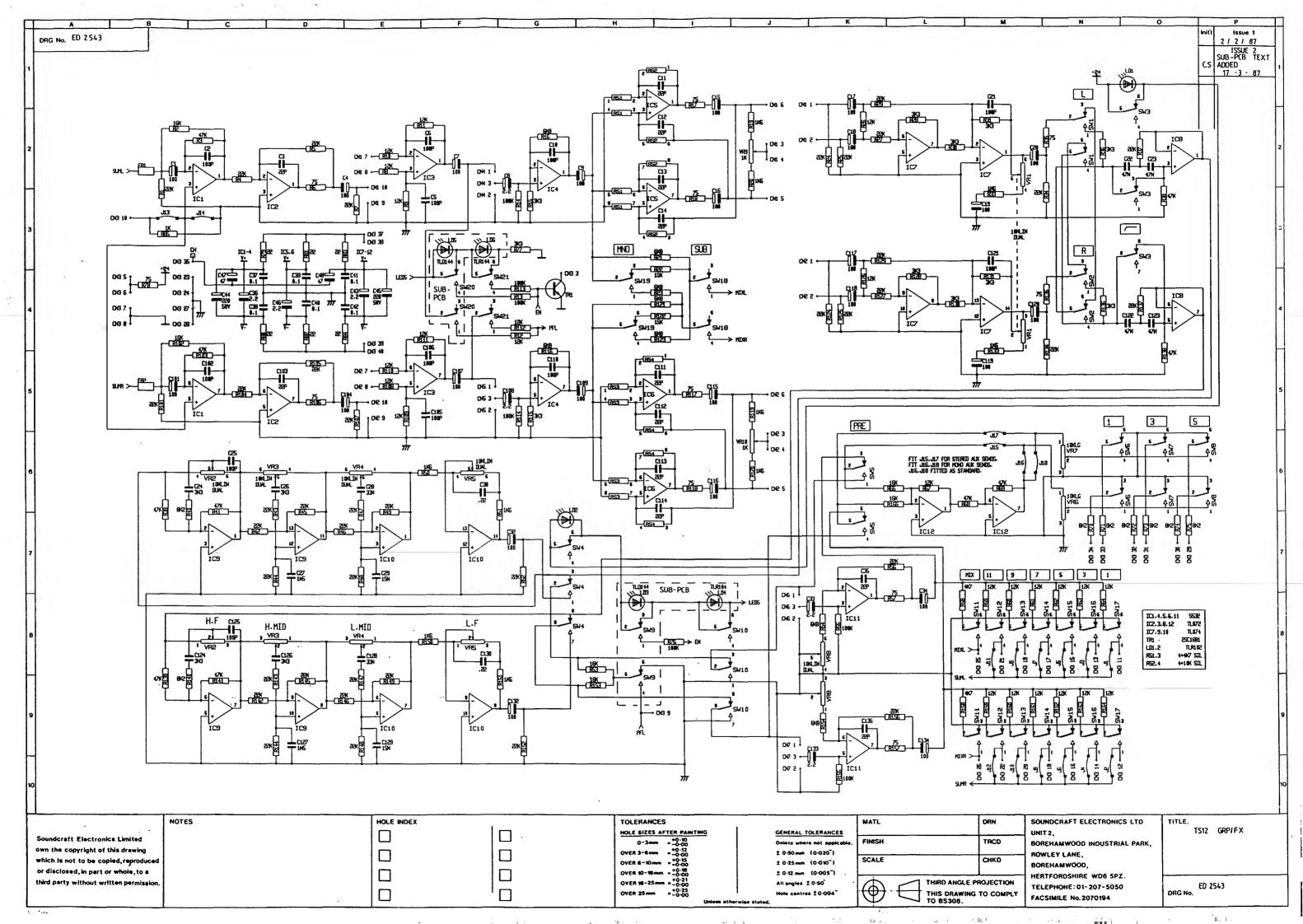


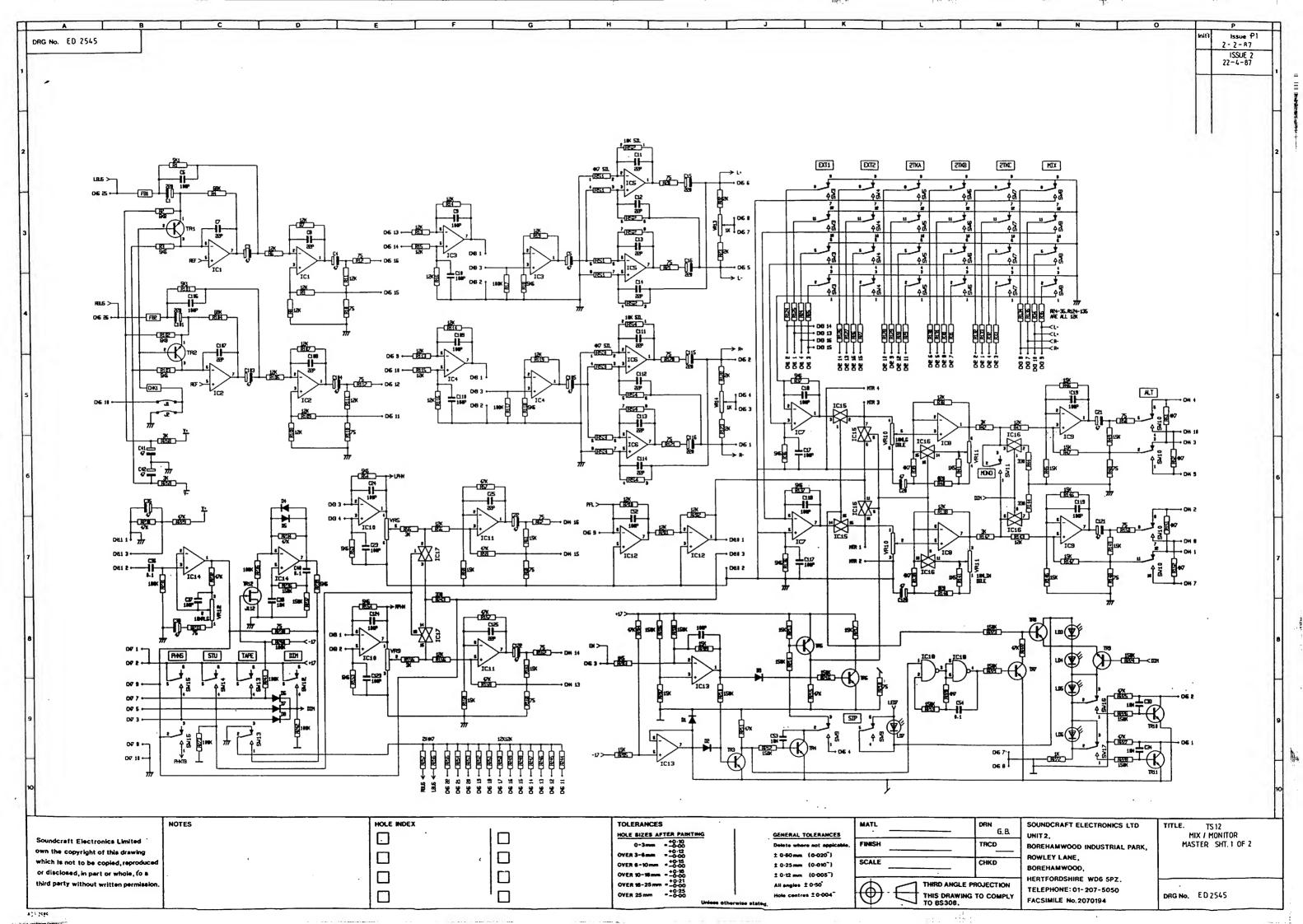


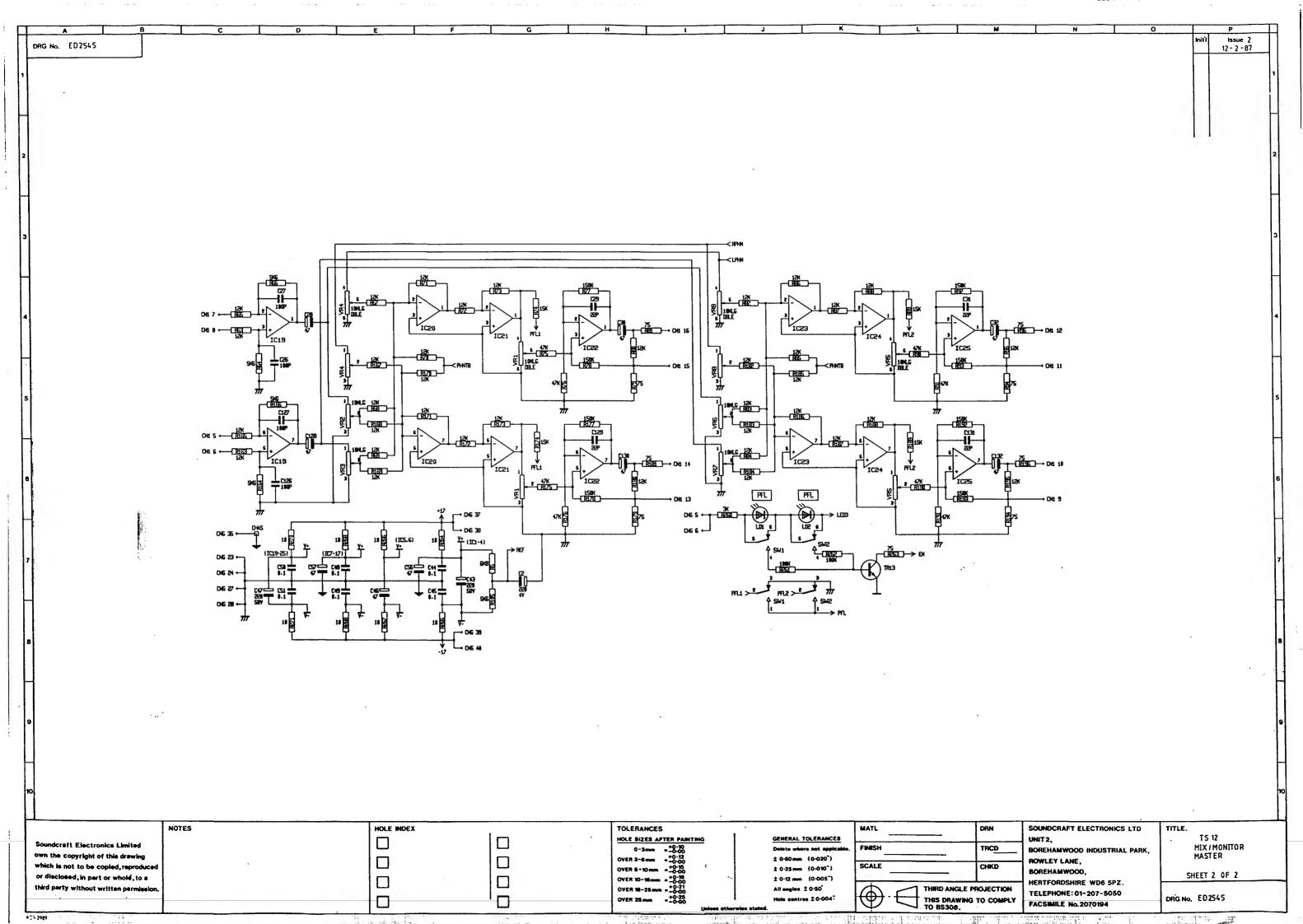


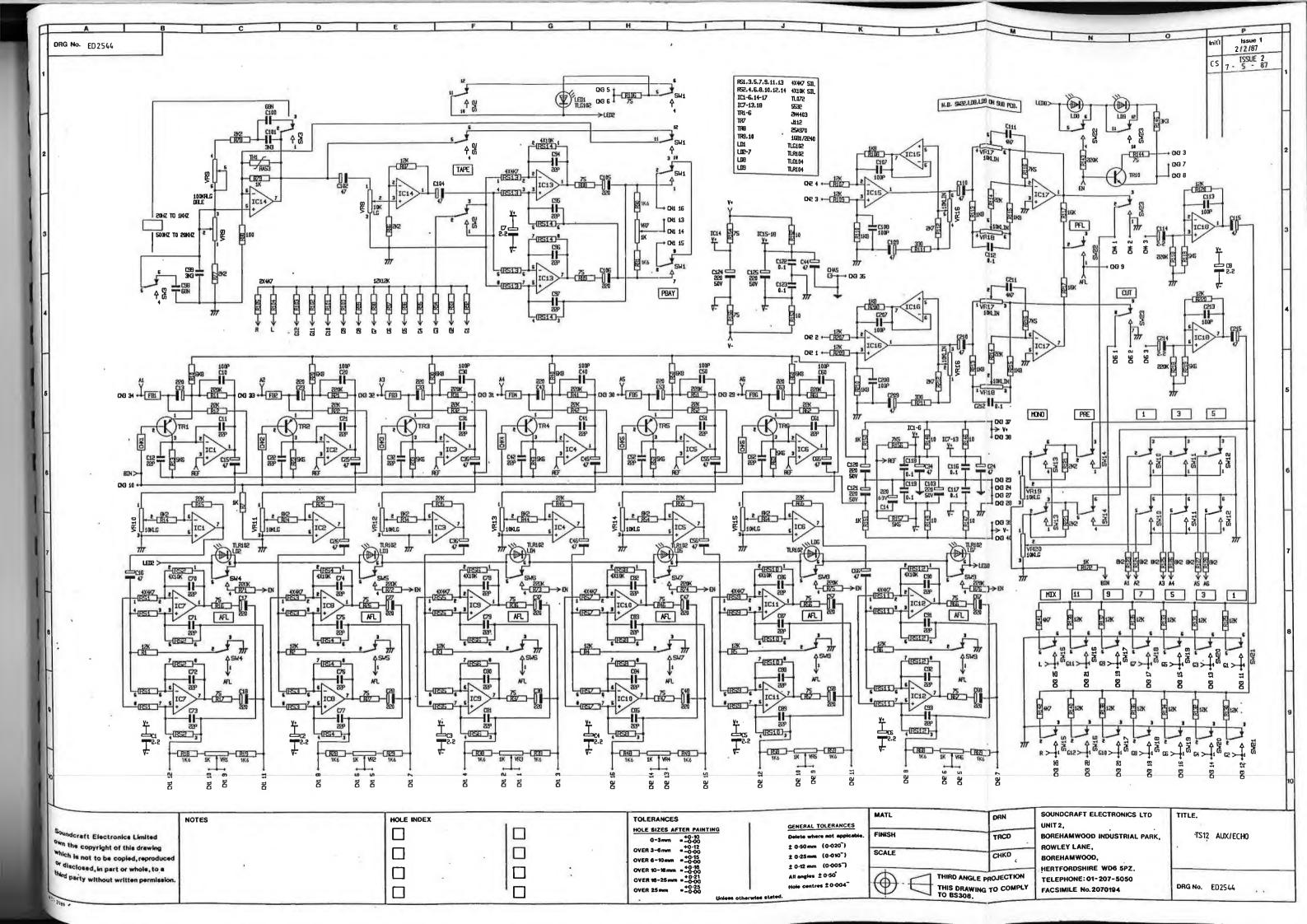


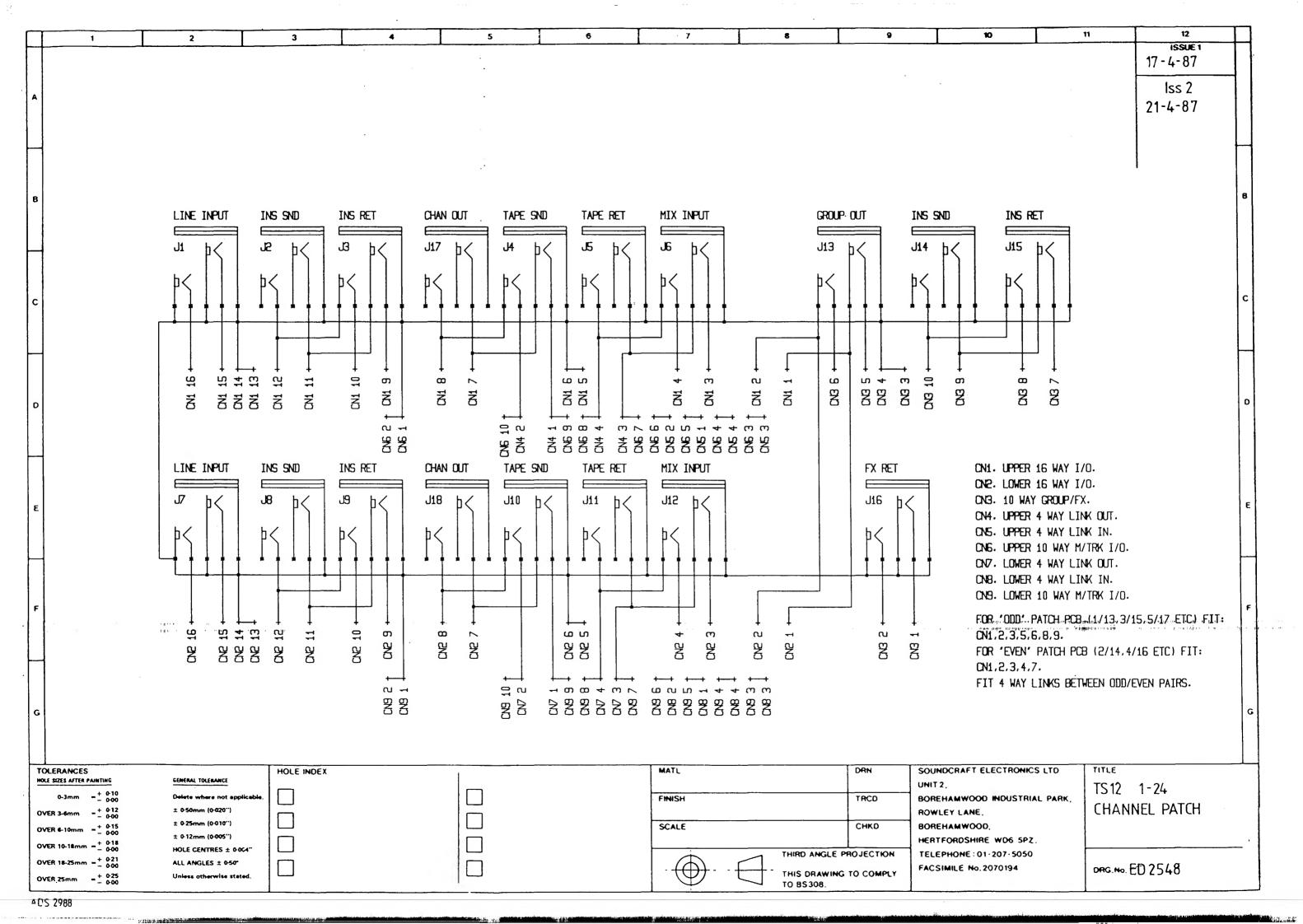


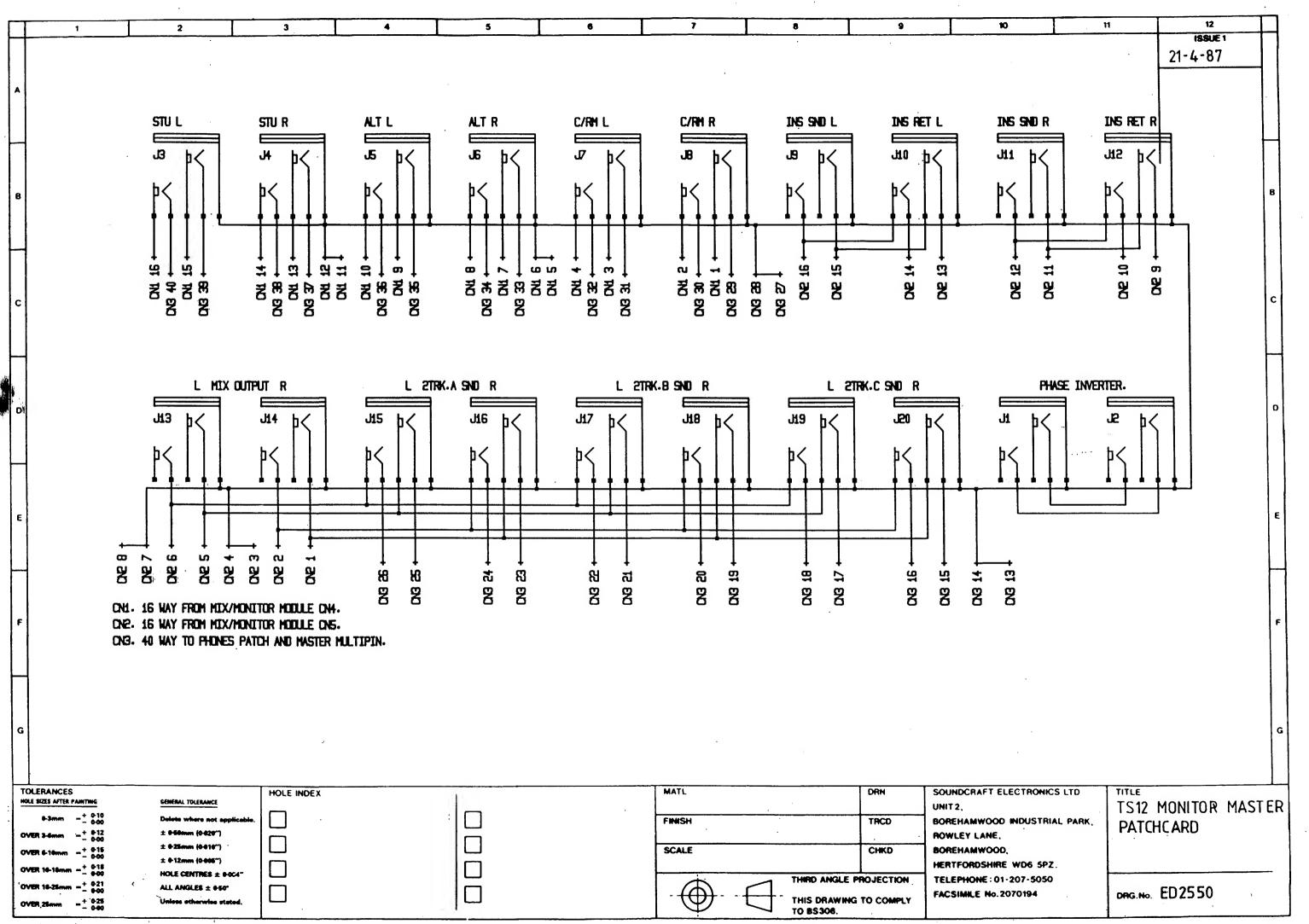




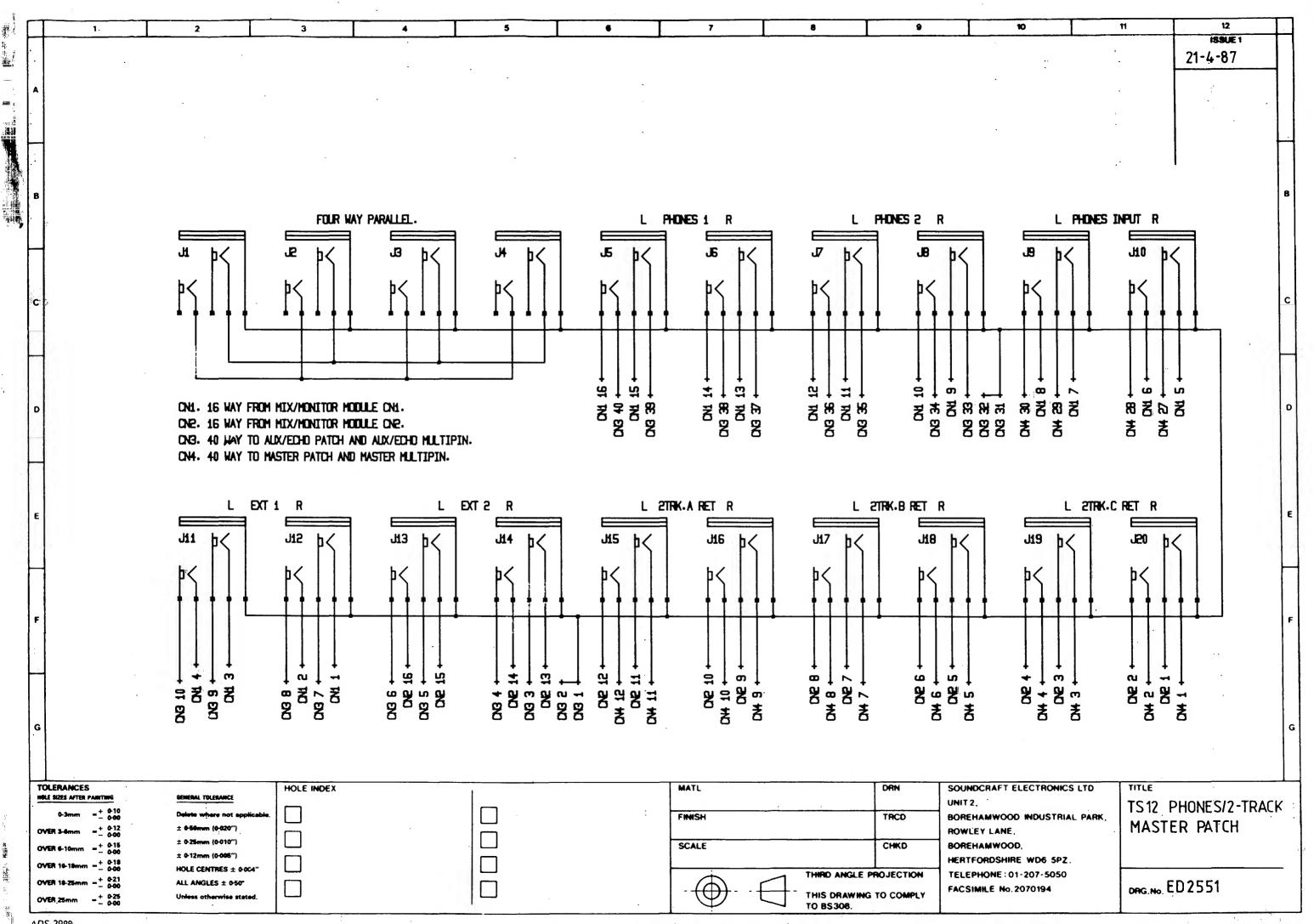




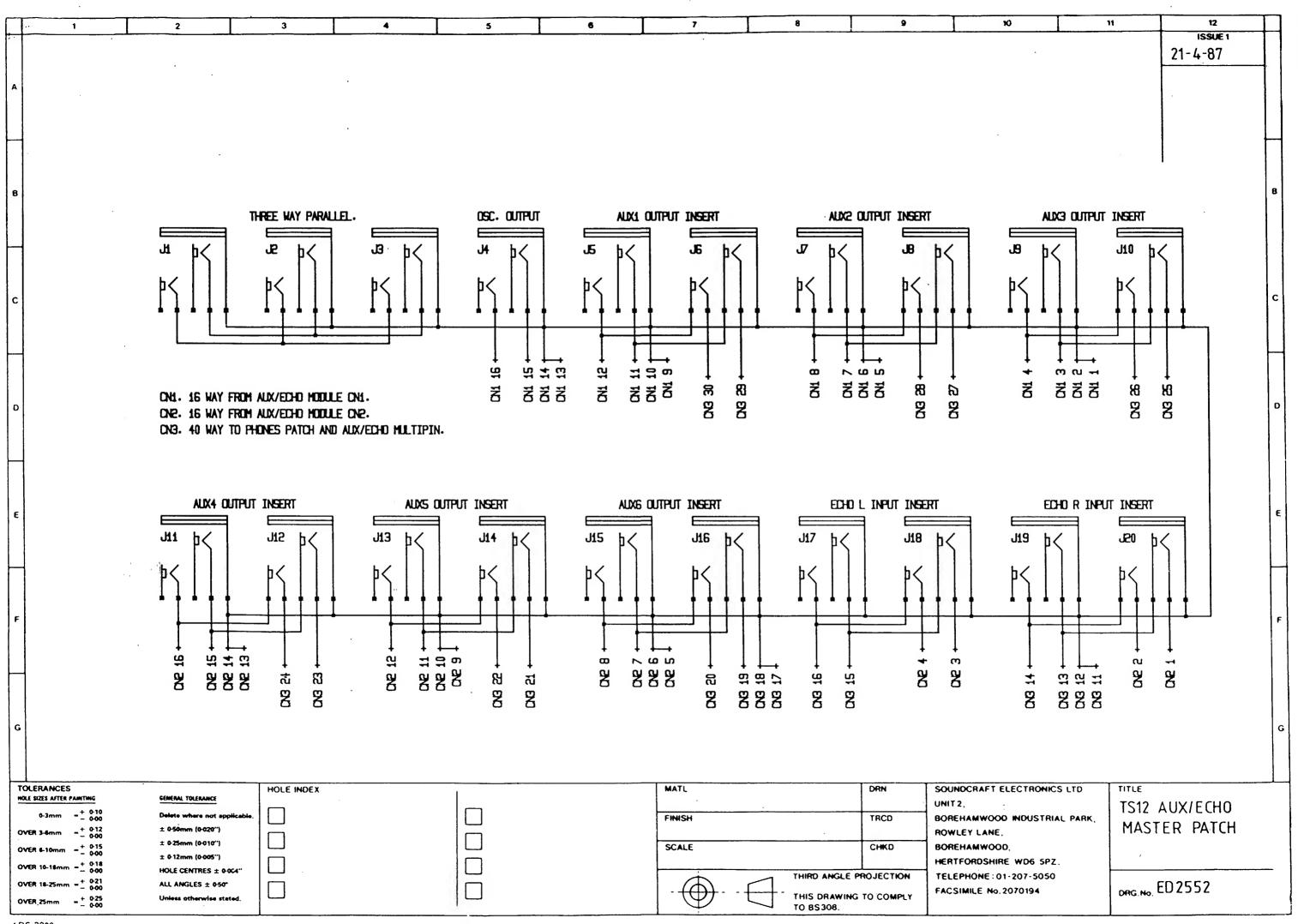


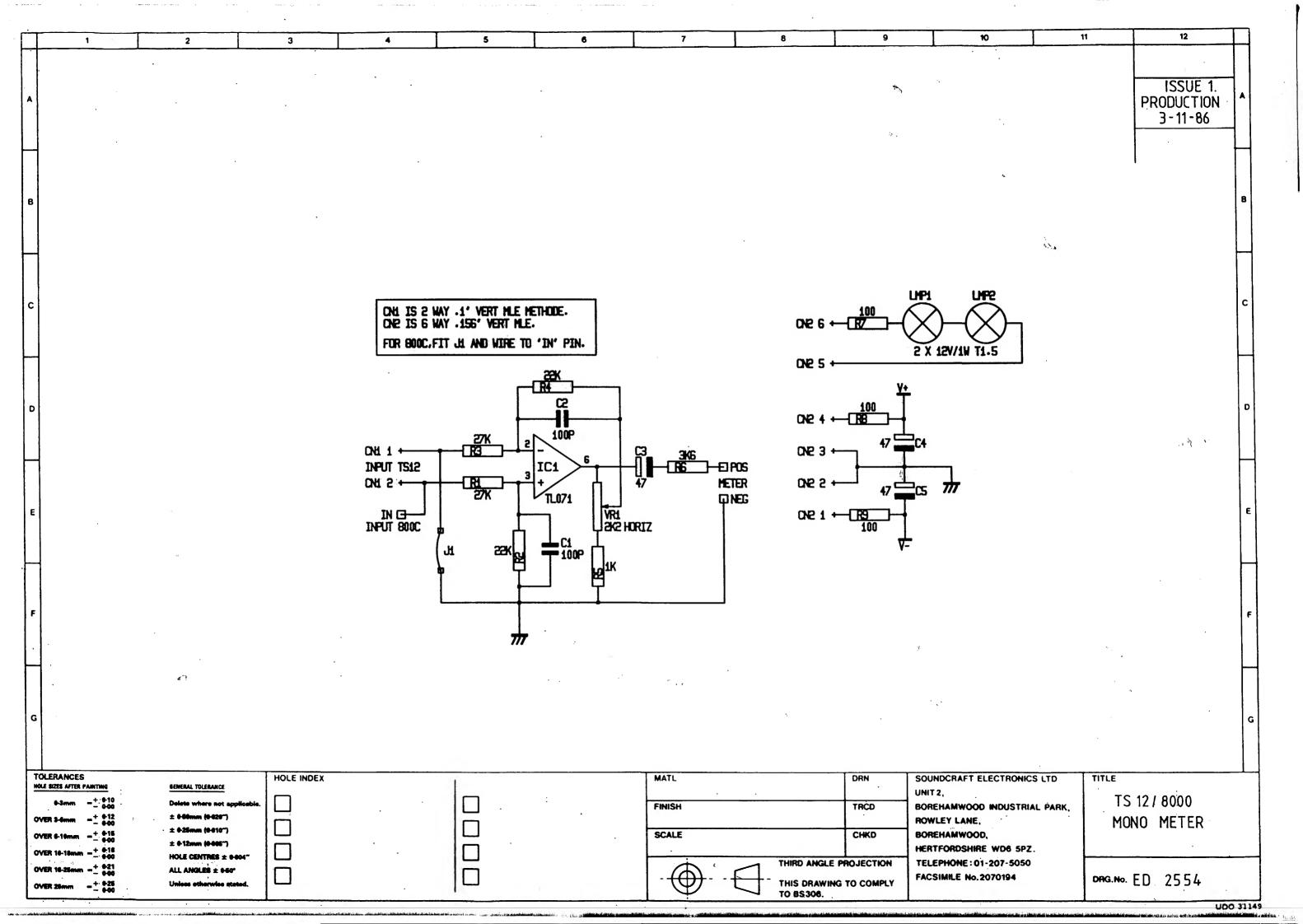


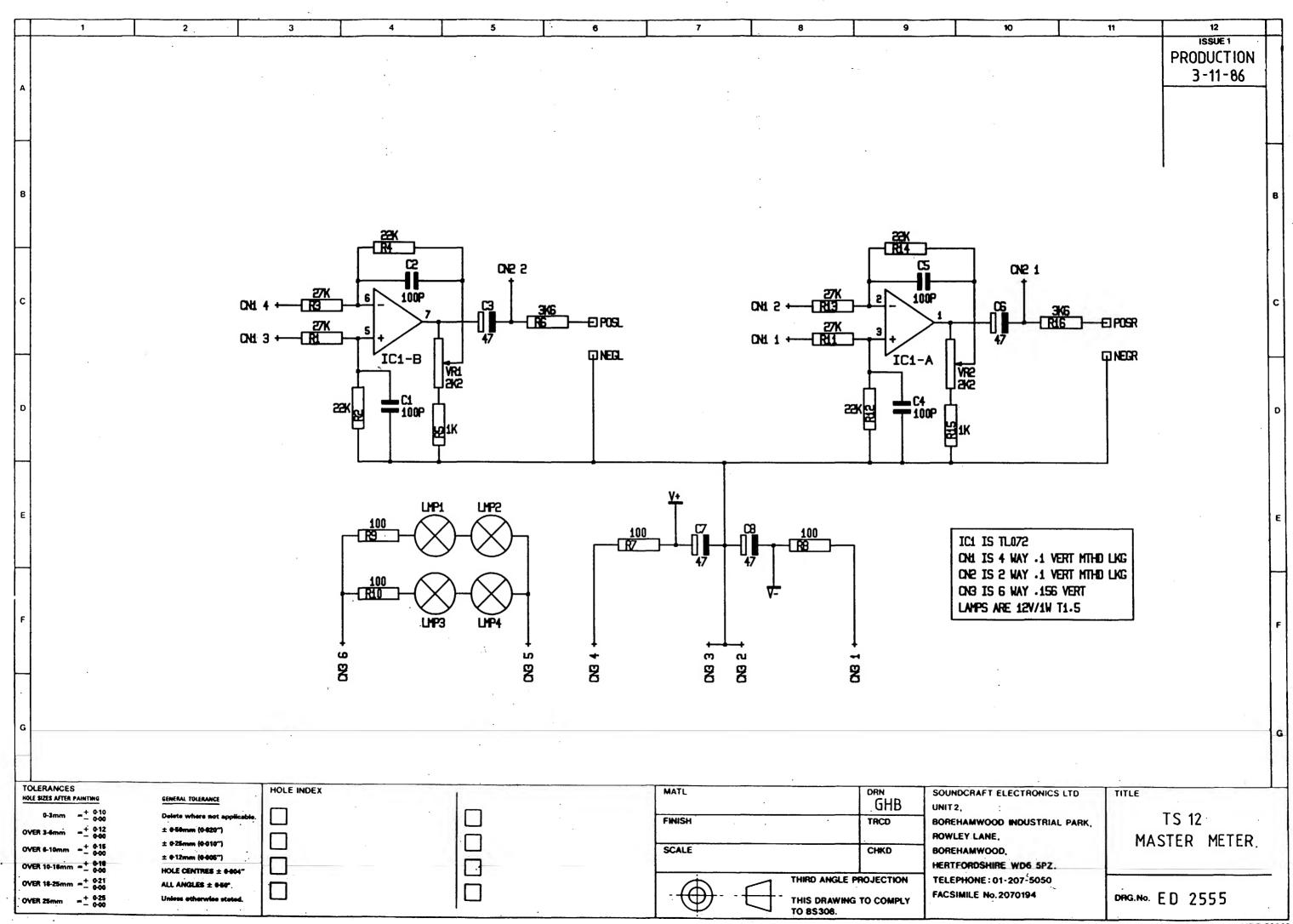
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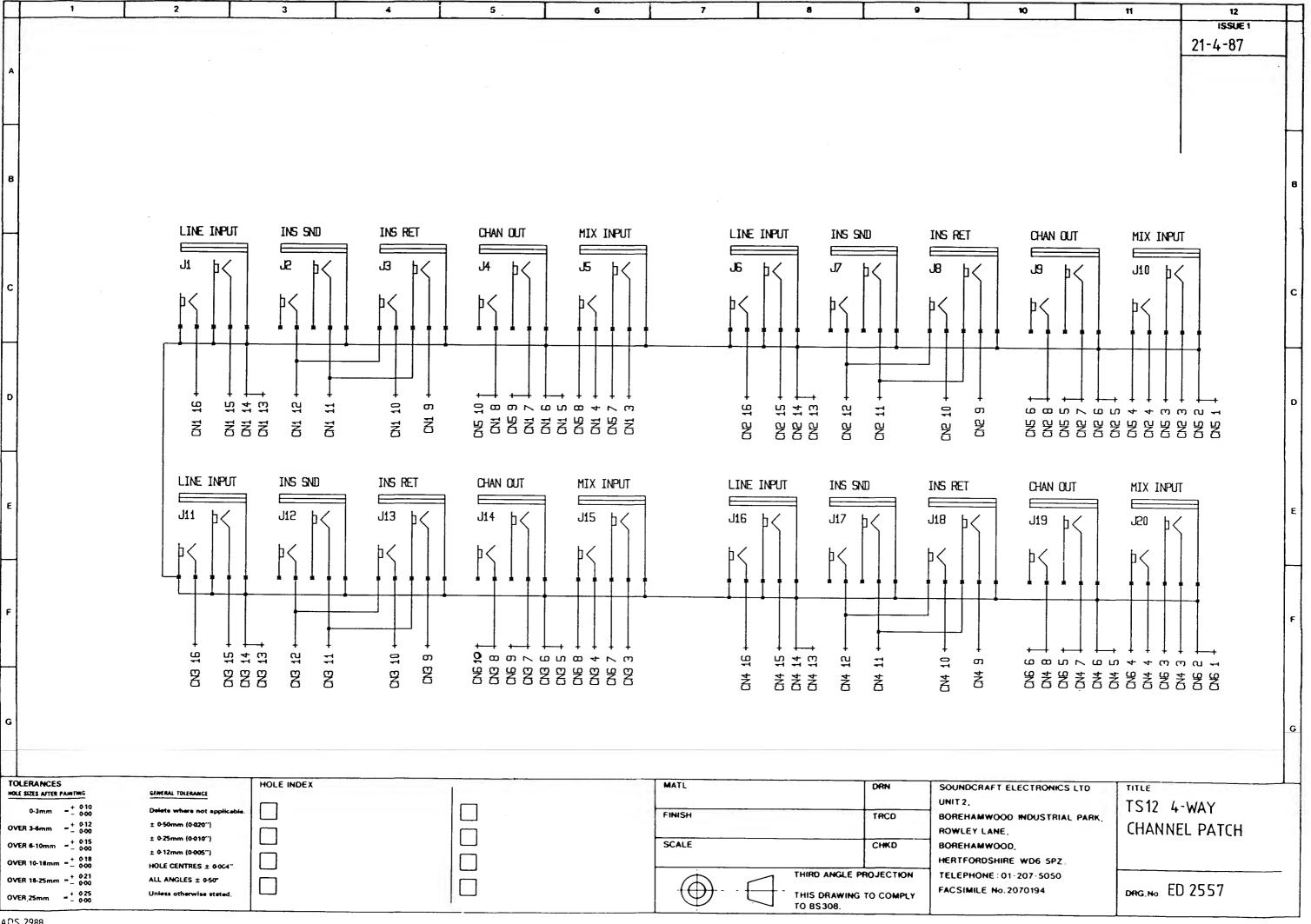


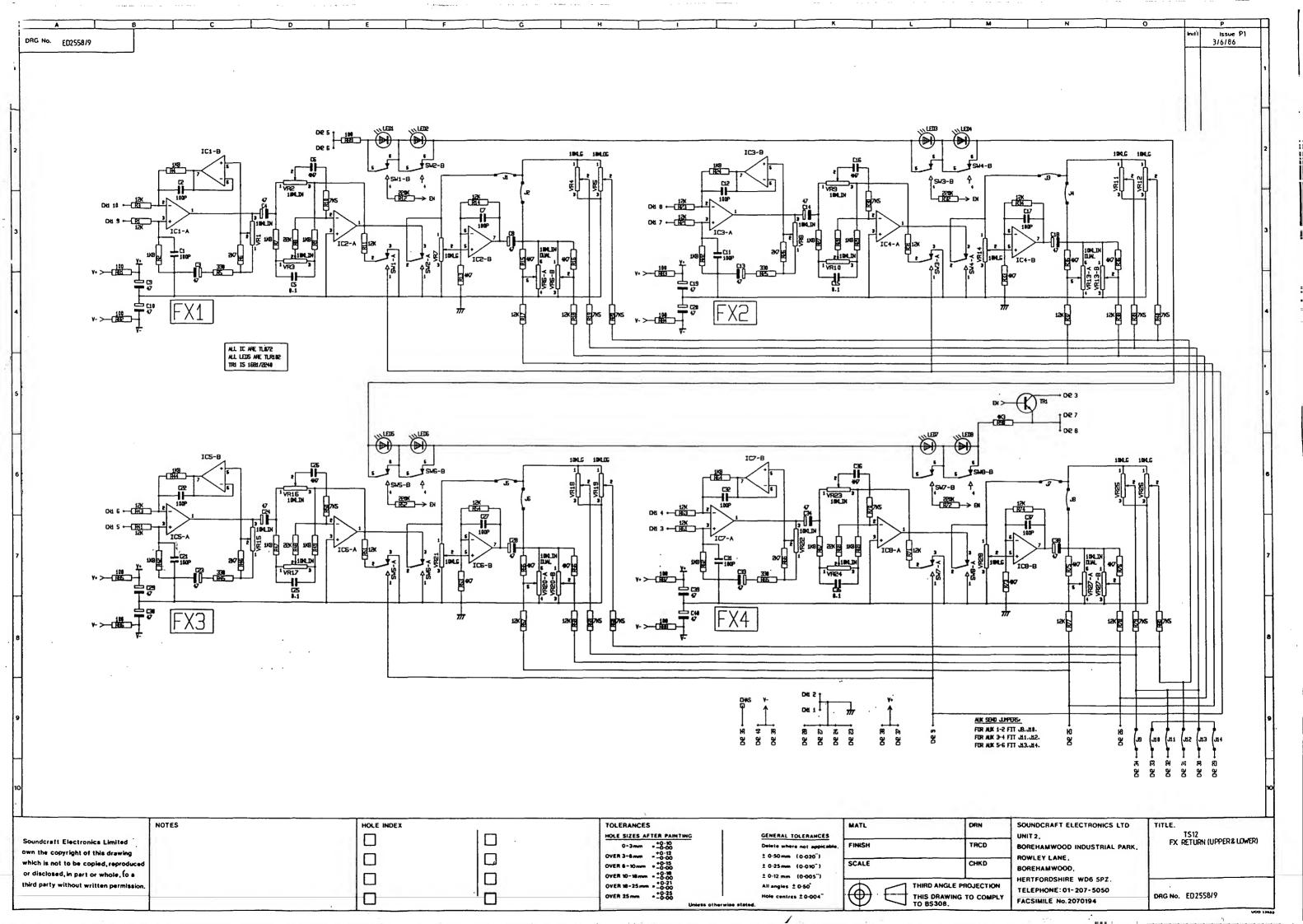
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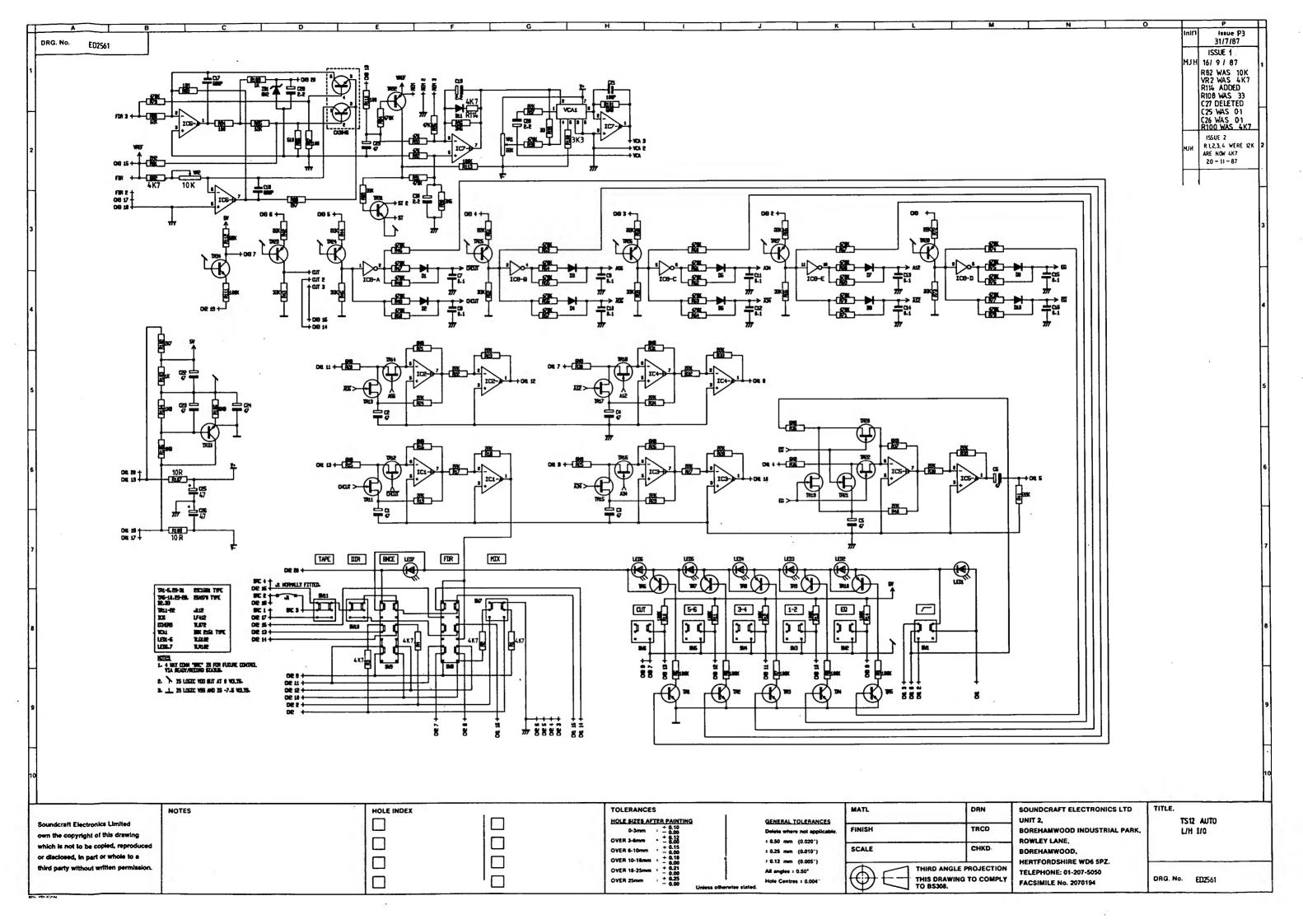


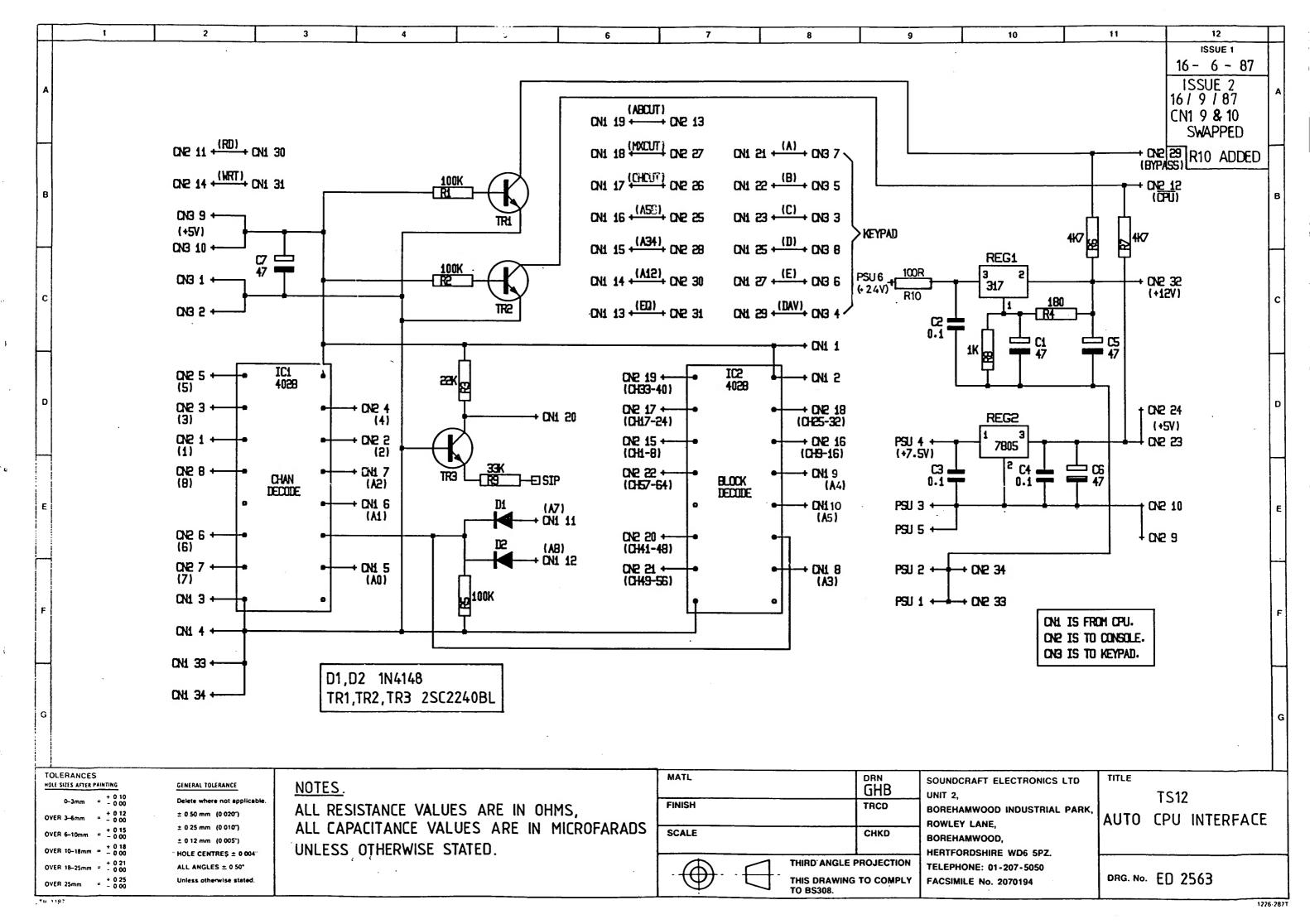


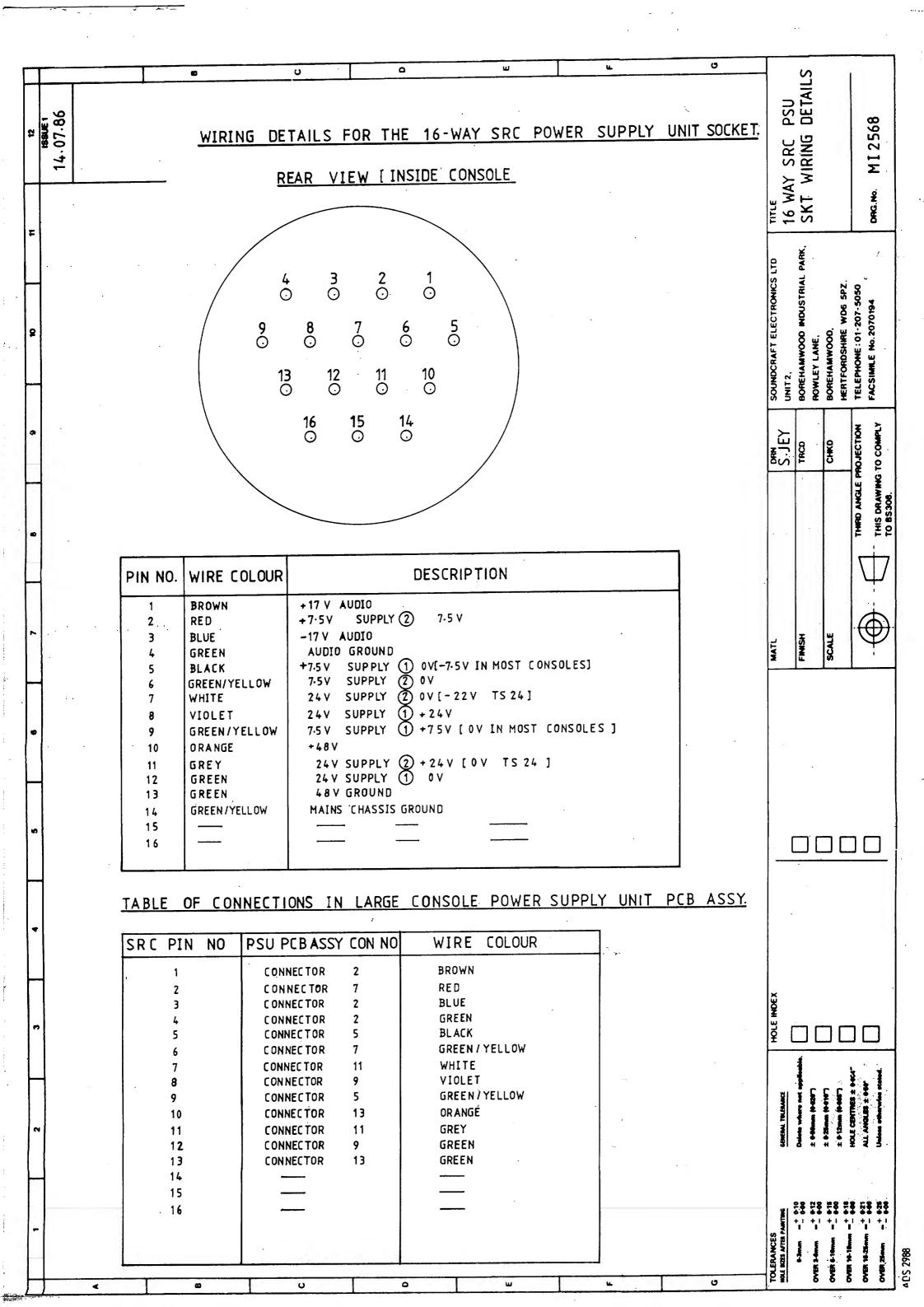


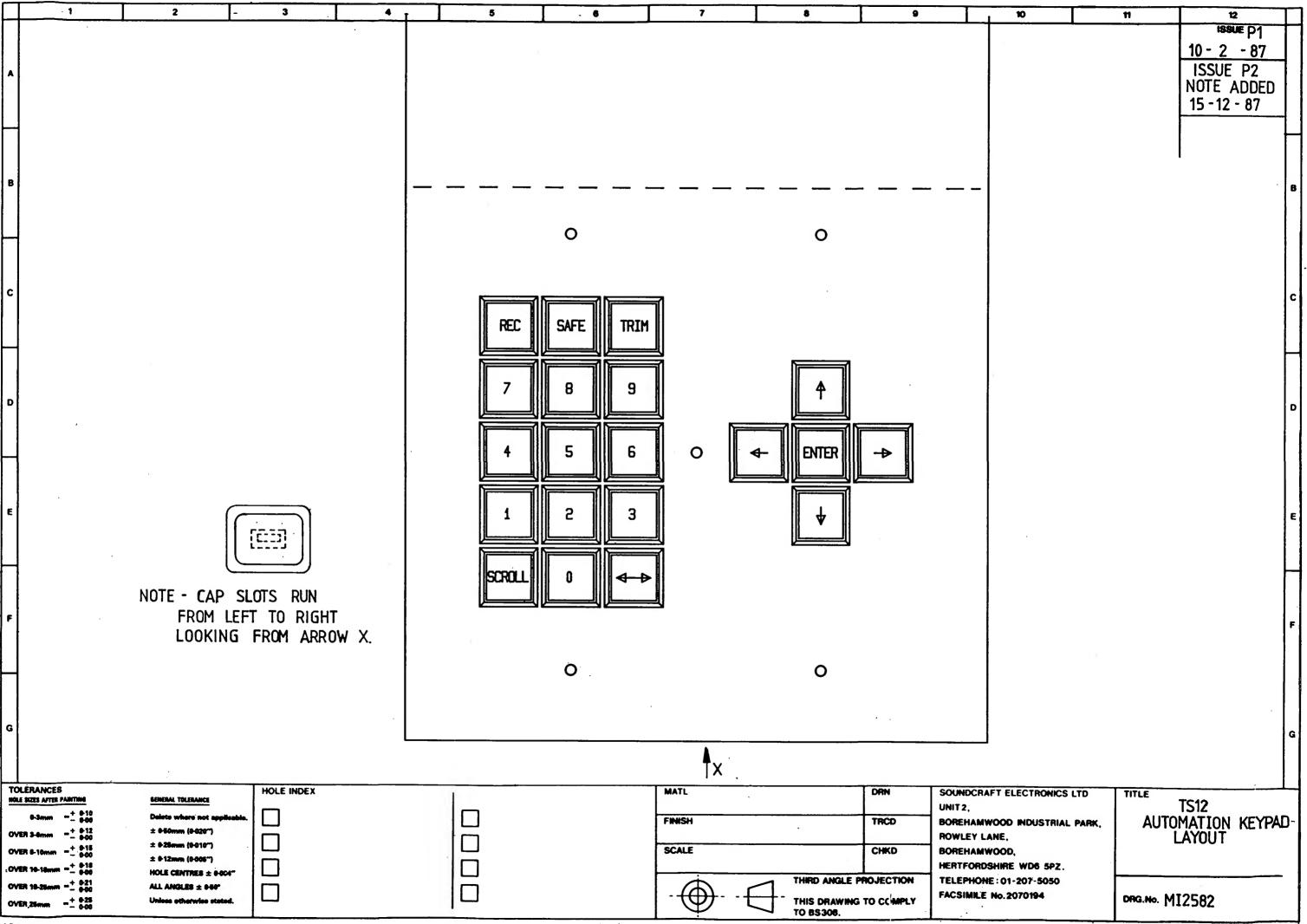


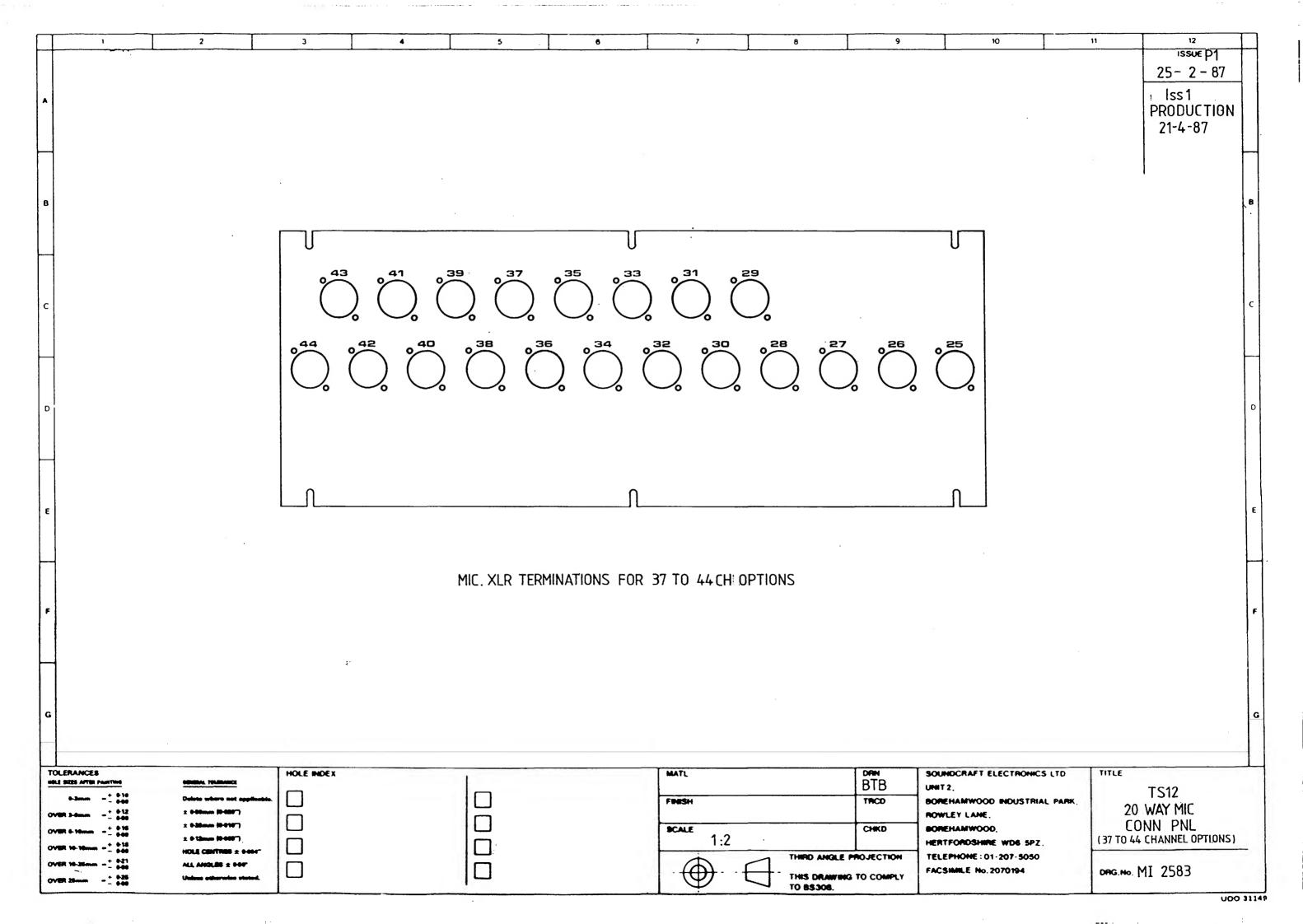


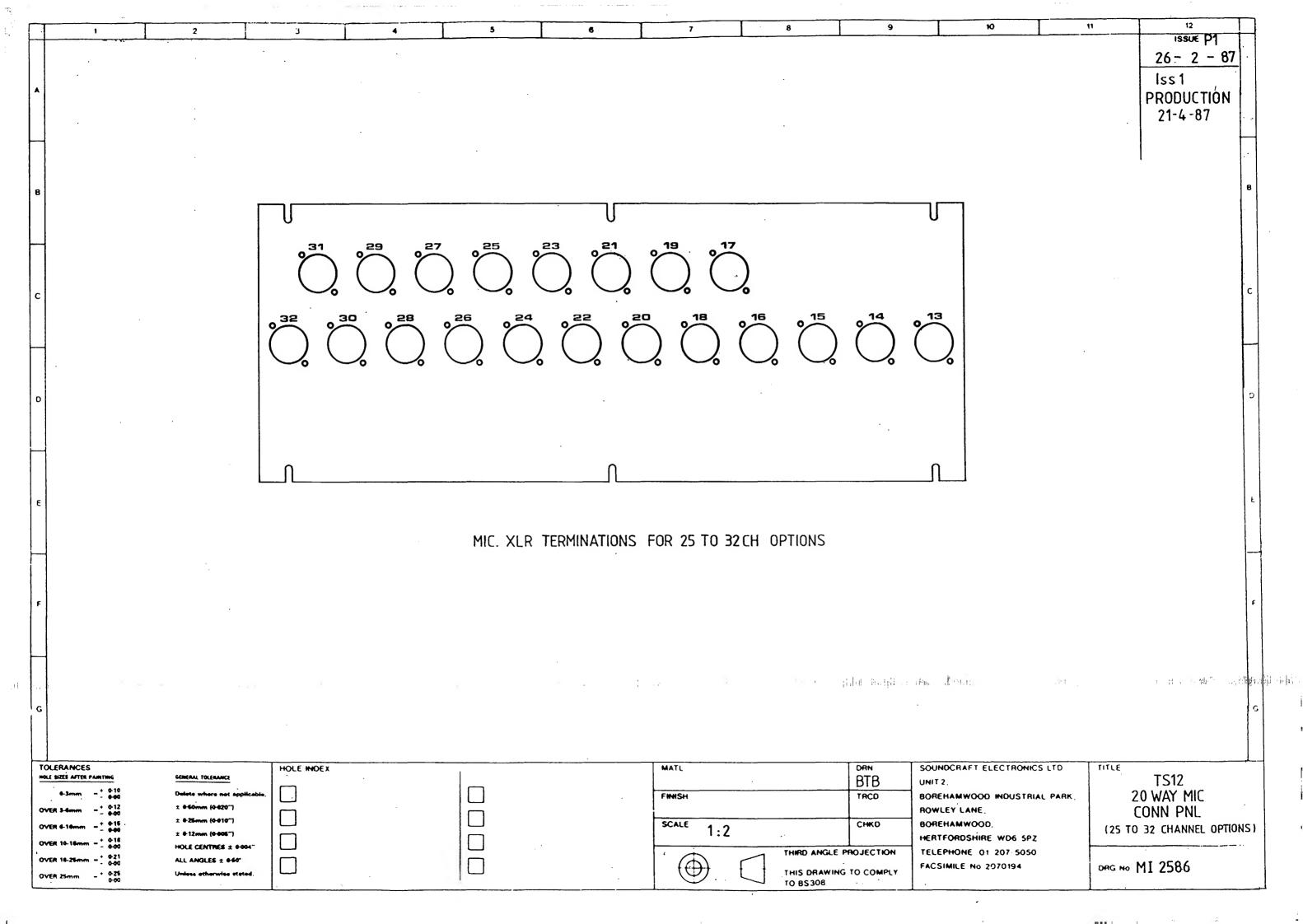


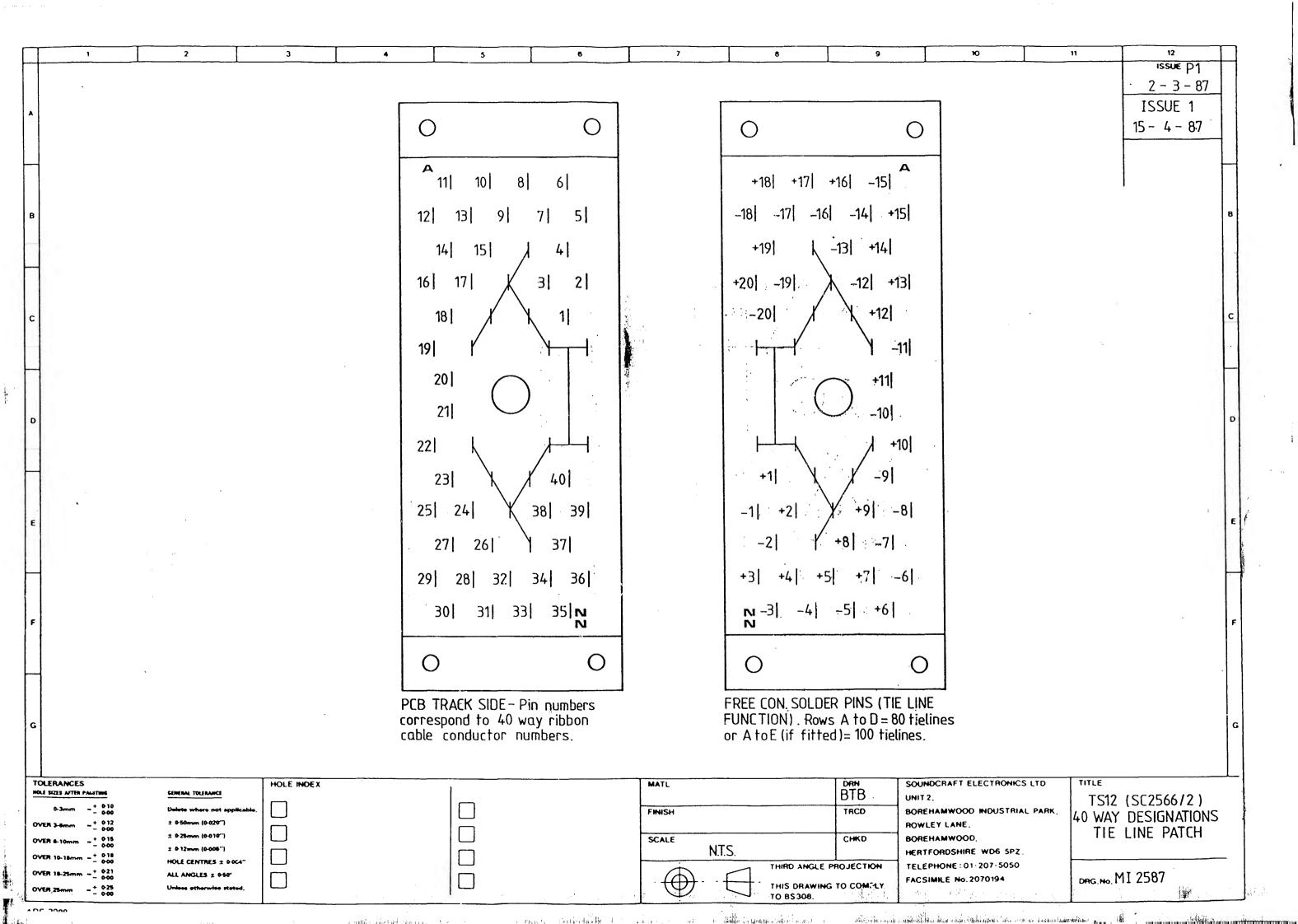


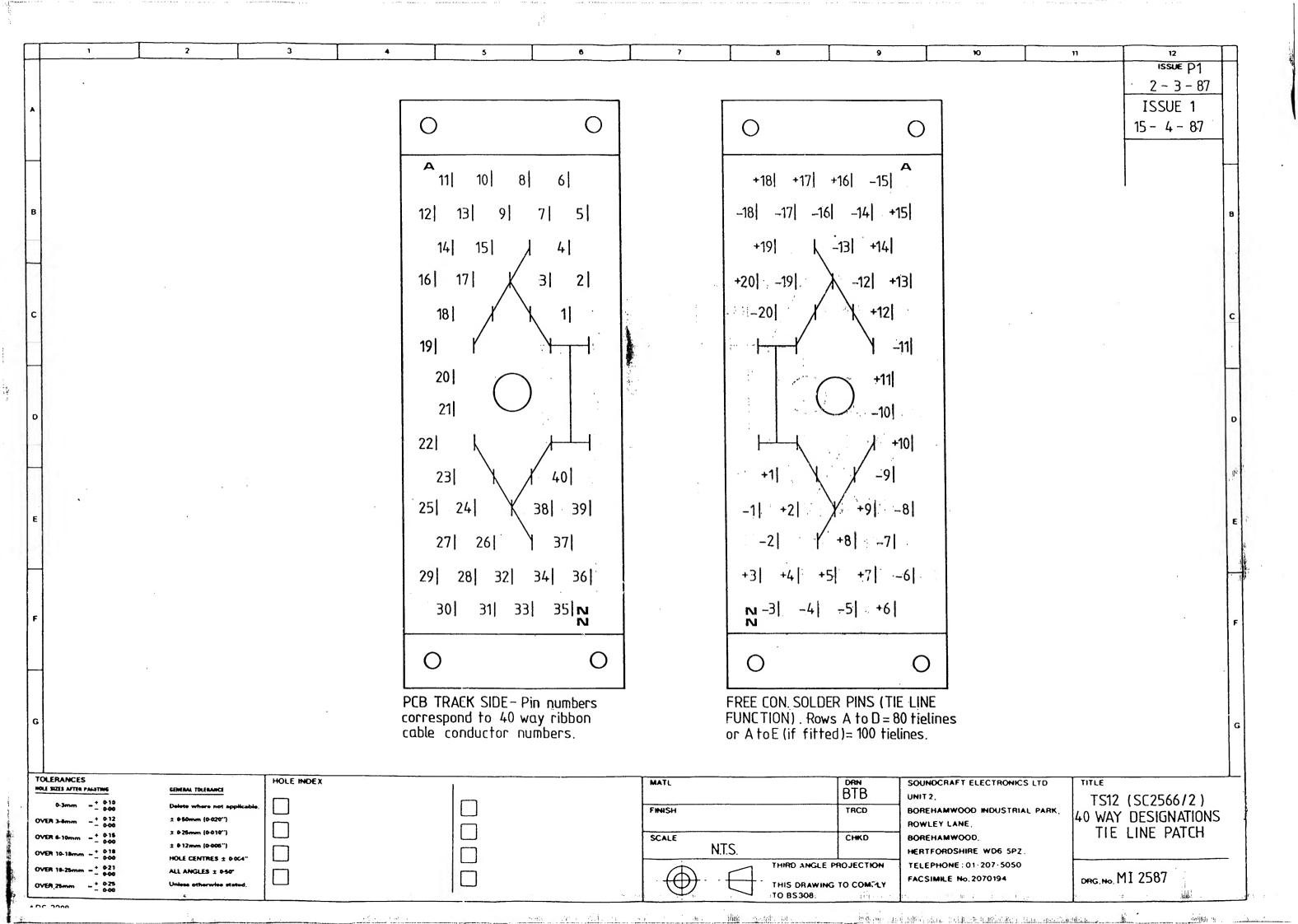


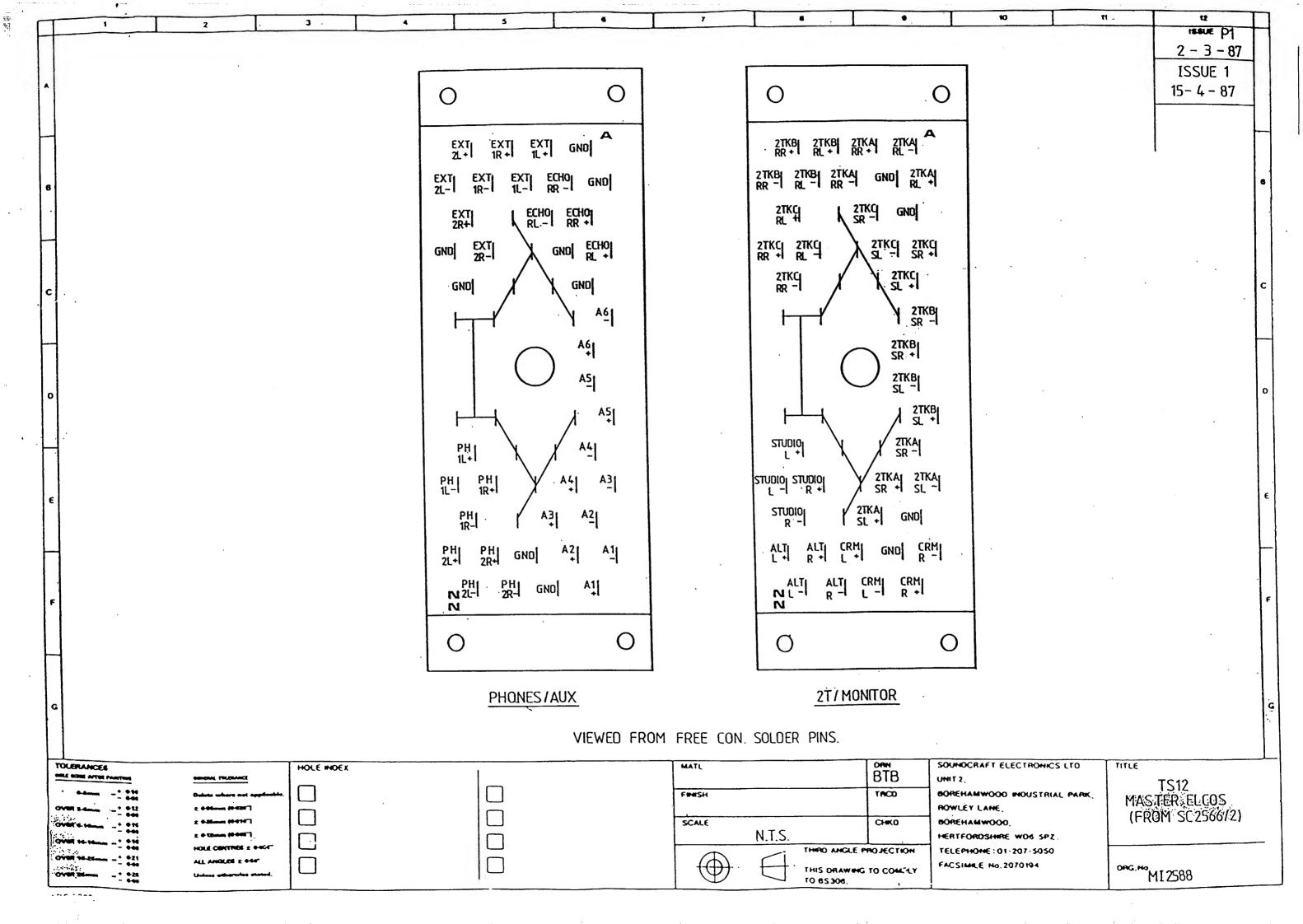


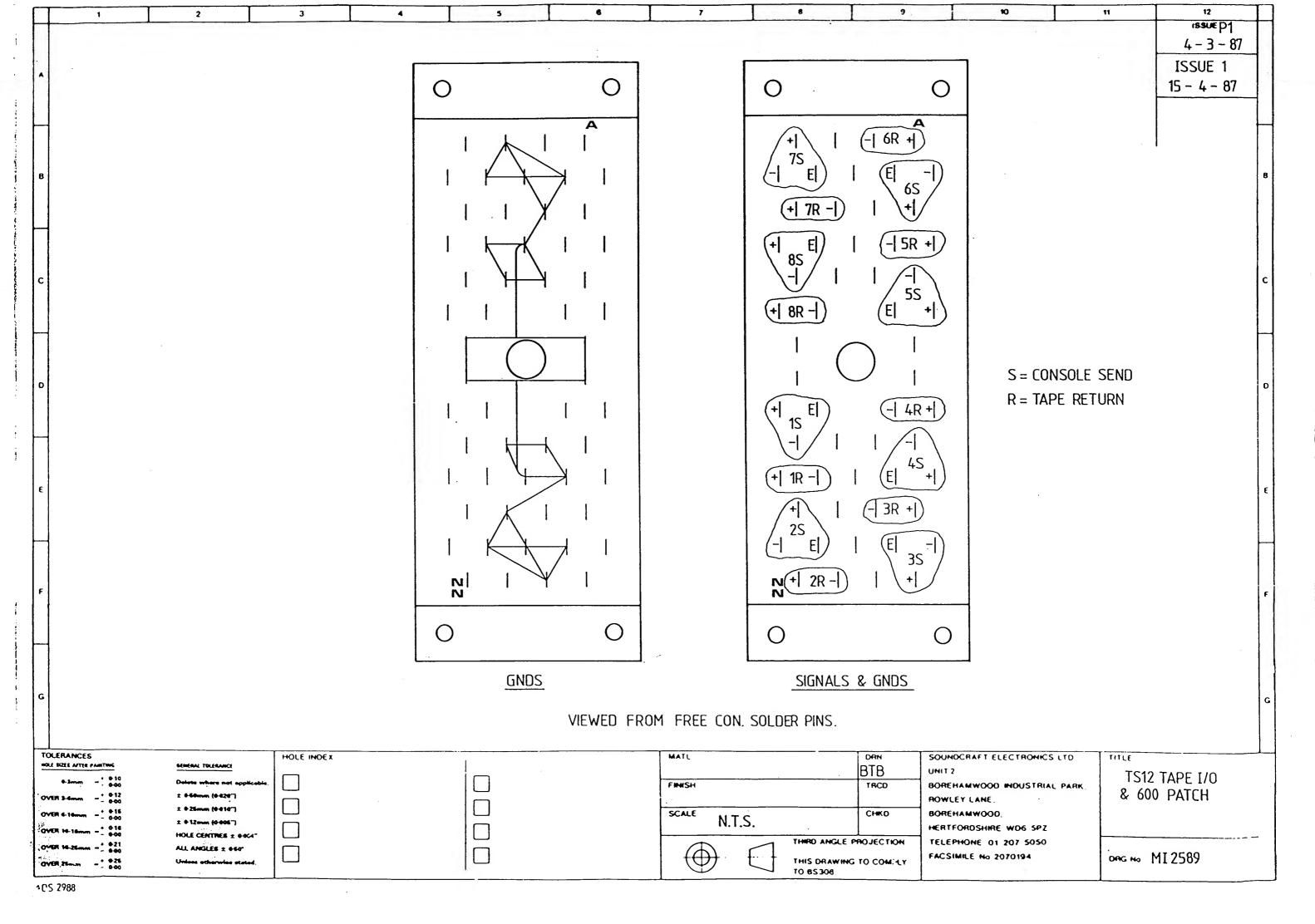












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TS12 ADC PATCHBAY MI2619

